ATTORNEY DOCKET NO. BANIK 2-73 PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

In re Application of: Somnath Banik, et al.

Serial No.: 09/514,489

Filed: February 29, 2000

For: SYSTEM AND METHOD FOR COMMUNICATING DATA OVER A RADIO FREQUENCY VOICE CHANNEL

Grp./A.U.: 2684

Examiner: Tu X. Nguyen

I hereby certify that this correspondence is being electronically filed with United States Patent and trademark Office on June 12, 2006

Marty Balko
(Printed or typed name of person signing the certificate)

/Marty Balko/
(Signature of the person signing the certificate)

Mail Stop Appeal Brief-Patents

ATTENTION: Board of Patent Appeals and Interferences

Sirs:

APPEAL BRIEF UNDER 37 C.F.R. §41.37

This is an appeal from a Final Rejection dated January 3, 2006, of Claims 1, 2, 4-9 and 11-20. The Appellants submit this Brief with the statutory fee of $500.00 as set forth in 37 C.F.R.§41.20(b)(2), and hereby authorize the Commissioner to charge any additional fees connected with this communication or credit any overpayment to Deposit Account No. 08-2395.

This Brief contains these items under the following headings, and in the order set forth below in aceordance with 37 C.F.R. §41.37(c)(1):

I. REAL PARTY IN INTEREST

The real party in intcrest in this appeal is the Assignee, Agere Systems Inc.

II. RELATED APPEALS AND INTERFERENCES

No other appeals or interferences will directly affect, be dirctly affected by, or have a bcaring on the Board's decision in this appeal.

III. STATUS OF THE CLAIMS

Claims 1-2, 4-9 and 11-22 are pending in this application and have been rejected under §102(e) or §103(a). Each of the pending claims are being appealed.

IV. STATUS OF THE AMENDMENTS

The present Application was filed on February 29, 2000, with Claims 1-20. The Appellants filed a first Request for Reconsideration on August 15, 2002, in response to a first Examiner's Action mailed May 24, 2002. The Examiner considered the Appellants' arguments moot in view of new grounds for rejections as asserted in the second Examiner's Action mailed on October 23, 2002. The Appellants then filed a first Amendment on January 16, 2003, that amended Claims 1 and 8 and canceled Claims 3 and 10. The Examiner again considered the Appellants' arguments moot in view of new grounds for rejections as asserted in the third Examiner's Action mailed on March 17, 2003. The Appellants filed a second Request for Reconsideration on June 5, 2003, that argued against the rejections in the third Examiner's Action. The Examiner then filed a fourth Examiner's Action on July 25, 2003, that again considered the Appellants' arguments moot in view of new grounds for rejection.

On October 23, 2003, the Appellants filed a second Amendment that again amended Claims 1 and 8. The Examiner considered the Appellants' arguments moot in view of new grounds for rejections as asserted in the fifth Examiner's Action mailed on January 5, 2004. In response, the Appellants filed a third Amendment on April 2, 2004, that added Claims 21-22. A sixth Examiner's Action was then filed on May 7, 2004, citing new grounds for rejections of all the pending claims. The Appellants responded with a third Request for Reconsideration on August 4, 2004, that argued

for the allowance of the pending claims. On November 3, 2004, the Examiner mailed a first Final Rejection rejecting all of the pending Claims 1-2, 4-9 and 11-22. The Appellants filed a fourth Request for Reconsideration on December 29, 2004, that the Examiner responded to on January 26, 2005, with a first Advisory Action maintaining the final rejections. The Appellants then filed a first Notice of Appeal on February 3, 2005, for Claims 1-2, 4-9 and 11-22 as filed with the second Amendment of April 2, 2004. On March 30, 2005, the Appellants filed a first Appeal Brief.

In view of the first Appeal Brief, the Examiner reopened prosecution and issued a seventh Examiner's Action on June 30, 2005, citing new references for rejections of all the pending claims. The Appellants responded by filing a fifth Request for Reconsideration on October 28, 2005, that traversed the rejection. On January 3, 2006, the Examiner mailed a second Final Rejection rejecting all of the pending Claims 1-2, 4-9 and 11-22. The Appellants filed a sixth Request for Reconsideration on February 23, 2006, that the Examiner responded to on March 7, 2006, with a second Advisory Action maintaining the final rejections. The Appellants then filed a second Notice of Appeal on April 3, 2006, for Claims 1-2, 4-9 and 11-22 as filed with the second Amendment of April 2, 2004. Additionally, the Appellants filed a Pre-Appeal Brief Request for Review. In response, a Notice of Panel Decision from Pre-Appeal Brief Review was mailed on May 10, 2006, indicating that the Appeal should proceed to the Board of Patent Appeals and Interferences. On May 31, 2006, the Examiner granted an interview to discuss the second Final Rejection. The rejection was maintained resulting in the present second Appeal Brief.

V. SUMMARY OF CLAIMED SUBJECT MATTER

The present invention is directed, in general, to a communication system and, more specifically, to a system and method of sending data over a voice channel. (*See* page 1, lines 4-6.) The present invention introduces the broad concept of sending packets of data during pauses or periods of silence that occur during voice interchanges. The present invention provides substantial utility by making efficient use of the communication channel and eliminates possible distortions in the voice transmission that may occur if data and voice are superimposed. (*See* page 4, lines 12-18.)

Independent Claim 1 is directed to a system for communicating data over a voice channel between a transmitter of a base station and a receiver of a handset of a cordless telephone including: (1) a silence detector, coupled to the transmitter, that identifies a pause in voice traffic that is to be transmitted over the voice channel and generates an interjection signal during the pause and (2) a data injector, coupled to the silence detector, that receives the interjection signal and responds by causing the transmitter to transmit data to the receiver over the voice channel. (*See* page 4, lines 4-11; page 5, lines 6-8; and page 15, lines 1-10 and 15-17.)

In one embodiment, a data transmitter system 210 and a data receiver system 220 are associated with a base station 110 and a handset 120 of a cordless telephone 100, respectively. (*See* page 10, lines 9-14 and Figures 1-2.) The data transmitter system 210 includes a silence detector 214 and a data injector 216 in addition to a base station antenna 211, a transmitter/receiver 212, a telephone line interface 213 and a data register 215. Voice and data information are accepted from the phone lines by the interface 213. Voice information is presented to the transmitter/receiver 212 for transmission to the handset 120 in the regular manner via the base station antenna 211. Data information is routed to the data register 215, for holding, from the interface 213 until it is

appropriate for it to be transmitted to the data receiver system 220. The silence detector 214, coupled to the transmitter/receiver 212, identifies a pause in voice traffic and generates an interjection signal during the pause. This interjection signal enables the data injector 216 during this pause, causing the transmitter/receiver 212 to transmit data to the data receiver system 220 over the voice channel. (*See* page 10, line 18 to page 11, line 9.)

Independent Claim 8 is directed to a method of communicating data over a voice channel between a transmitter of a base station and a receiver of a handset of a cordless telephone, including: (1) identifying a pause in voice traffic that is to be transmitted over the voice channel and (2) responding to the pause by causing the transmitter to transmit data to the receiver over the voice channel. (*See* page 12, lines 17-22, page 17, lines 1-6 and 11-13 and Figure 3.)

In one embodiment illustrated in Figure 3, the transmitter receives information from phone lines in a step 310. The received information is monitored as to whether it is voice or data in a step 315. Data information is buffered and a pause in voice traffic is also identified in the step 315. (*See* page 13, lines 3-9 and Figure 3.)

In a first decision point denoted "Silence Frames ?" in a step 320, the decision is made as to whether there is a pause in the voice information indicating a silence frame. If the decision is YES and a pause in voice information has been identified, the data information is injected into an interstice in the voice information in a step 325. Then, in a step 330, the data information is transmitted to the receiver for display. (*See* page 13, line 10 to page 14, line 3 and Figure 3.)

Independent Claim 15 is directed to a cordless telephone, including: (1) a base station transceiver, (2) a handset transceiver, wherein the base station and handset transceivers are cooperable to establish a voice channel therebetween, (3) a silence detector, coupled to the base

station transceiver, that identifies a pause in voice traffic that is to be transmitted over the voice channel and generates an interjection signal during the pause and (4) a data injector, coupled to the silence detector, that receives the interjection signal and responds by causing the base station transceiver to transmit data to the receiver over the voice channel. (*See* page 19 lines 1-12 and Figures 1-2.)

In one embodiment illustrated in Figure 1, a cordless telephone 100 includes a base station transceiver 110 and a handset transceiver 120. The base station transceiver 110 is connected to a telephone line 115 and includes a base station antenna 111 and a handset cradle 112. (*See* page 8 lines 1-21 and Figure 1.) The cordless telephone 100 of Figure 1 identifies a pause in voice traffic that is to be transmitted over a voice channel and generates an interjection signal during the pause. The injection signal then causes the transmitter to transmit data to the receiver over the voice channel. (*See* page 9 lines 1-8 and Figure 1.)

Regarding the dependent claims, the voice traffic may be analog voice traffic. (*See* page 4, line 21 to page 5, line 5; page 9, lines 8-12; and page 12, line 21 to page 13, line 2.) Additionally, the data may be caller identification data or menu item selection data. (*See* page 5, lines 12-23; page 9, line 18 to page 10, line 8; and page 11, line 18 to page 12, line 2.) The transmitter may also transmit the voice traffic in frames. (*See* page 6, lines 1-6; page 12, lines 3-6; and page 13, lines 10-12). Furthermore, a may pause in voice traffic may be identified by comparing a peak energy of the voice traffic to a noise floor reference. (*See* page 6, lines 7-9 and page 12, lines 10-14.)

Regarding dependent Claims 21-22, the voice traffic and the data may be received from a telephone line coupled thereto. The base station transceiver 110 is connected to a telephone line 115. (*See* page 8, lines 4-8 and Figure 1.) The data transmitter system 210 includes a base station antenna

211, a transmitter/receiver 212, a telephone line interfacc 213, a silence detector 214, a data register 215 and a data injector 216. Voice and data information are accepted from the phonc lincs by the interface 213. (*See* page 10, lines 18-22 and Figure 2.)

VI. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

The issue presented for consideration in this appeal is whether Claims 1-2, 4-6, 8-9, 11-13, 15-19 and 22, as rejected by the Examiner, are patentably nonobvious in accordance with 35 U.S.C. §103(a) over U.S. Patent No. 6,044,266 to Kato ("Kato") in view of U.S. Patent No. 5,960,357 to Kim ("Kim"). The second issue presented for consideration in this appeal is whethcr Claims 7, 14 and 20, as rejected by the Examiner, are patentably nonobvious in accordance with 35 U.S.C. §103(a) over Kato and Kim in view of U.S. Patent No. 6,301,287 to Wallcy, *et al.*, ("Walley").

VII. APPELLANTS' ARGUMENT

The inventions set forth in independent Claims 1, 8 and 15 and the respective dependcnt claims are not obvious over the references on which the Examiner relies.

Rejection undcr 35 U.S.C. 103(a) over Kato in view of Kim

A. Rejection of Claims 1, 8 and 15

The Examiner has rejected Claims 1, 8 and 15 under 35 U.S.C. §103(a) as being obvious over Kato in vicw of Kim. The Appellants rcspectfully disagree since Kato and Kim both fail to tcach or suggest identifying a pause in voice traffic that is to be transmitted over a voice channel and

responding to the pause by causing a base station transmitter to transmit data to a cordless telephone receiver over the voice channel as recited in independent Claims 1, 8 and 15.

Kato discloses a mobile packet data station b that monitors a voice path between a mobile voice station c and a base station to identify silent periods in the voice path. During the silent periods, the mobile packet data station transmits data therefrom to the base station. (*See* the Abstract; column 4, lines 6-26; and Figure 1.) Kato, therefore, teaches the mobile packet data station transmits data to the base station but does not teach or suggest the base station transmits data to the mobile voice station. This is clearly evident from Figure 1 of Kato that illustrates the packet transmission path f going from the mobile packet data station to the base station. (*See* column 4, lines 20-26.) The schematic timing diagrams of Kato also indicate that data packets are transmitted from the mobile packet data station to the base station. (*See* Figures 6B, 7B and 8B.) Kato provides no teaching or suggestion that data is transmitted **from** the base station **to** the mobile voice station. Even when voice bursts do occur between the mobile voice station and the base station, the mobile packet data station will look for another channel to transmit data packets to the base station. (*See* column 3, lines 4-10.)

The Examiner points to the second embodiment of Kato to indicate that the present invention can also be used on cordless telephones that do not have Voice-Operated Transmission (VOX) control. (*See* the second Advisory Action citing column 8, lines 40-45 and column 9, lines 35-36 of Kato.) The second embodiment differs from the first embodiment of Kato by being usable with a base station that does not use VOX control since the second embodiment adjusts a base station so that it can perform VOX control. (*See* column 8, lines 36-45 and column 9, lines 35-38.) Due to the

adjustment, down-link packet data communication can be performed by employing a VOX controlled channel. (*See* column 8, lines 42-45.)

Nevertheless, Kato, including the second embodiment, does not teach or suggest that the base station transmits data to the mobile voice station in response to an identified pause in voice traffic. On the contrary, "down-link packet data communication" as stated on line 42 of column 8, refers to when the data is transmitted from the mobile packet data station to the base station, not the path of transmission. (*See* column 4, lines 48-49 and Figure 5.) Even in the second embodiment, Kato teaches the **mobile packet data station** transmits data to **the base station** during silent periods. (*See* column 8, lines 45-46; column 9, lines 24-35; column 10, lines 2-23; and Figure 7B.) As such, Kato does not teach or suggest identifying a pause in voice traffic that is to be transmitted over the voice channel and responding to the pause by **causing a base station transmitter to transmit data to a cordless telephone receiver over the voice channel** as recited in independent Claims 1, 8 and 15. Thus, Kato fails to teach or suggest each element of independent Claims 1, 8 and 15.

Kim has not been cited to cure the above deficiencies of Kato but has been relied upon to teach a silence detector coupled to a transmitter. (*See* Examiner's second Final Rejection, pages 3-4.) Additionally, Kim does not eure the above deficiencies of Kato but instead is directed to executing an automatic calling function in a cordless telephone system including a fixed base station and a flip-type remote handset. (*See* column 1, lines 15-19.) The Applicants do not find any teaching or suggestion where Kim discloses the fixed based station transmits data to the flip-type remote handset in response to an identified pause in voice traffic therebetween. As such, the cited combination of Kato and Kim does not teach each and every element of independent Claims 1, 8 and 15.

Since the cited combination of Kato and Kim does not teach or suggest all of the elements of independent Claims 1, 8 and 15, the cited combination does not establish a *prima facie* case of obviousness of independent Claims 1, 8 and 15. Thus, Claims 1, 8 and 15 are not unpatentable in view of Kato and Kim. Accordingly, the Appellants respectfully request the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 1, 8 and 15 and issue allowance thereof.

B. Rejection of Claims 2, 9 and 16

The Examiner has rejected Claims 2, 9 and 16 under 35 U.S.C. §103(a) for being obvious over Kato in view of Kim. The above argument establishing that the combination of Kato and Kim does not render obvious the inventions of independent Claims 1, 8 and 15 is incorporated herein by reference. Dependent Claims 2, 9 and 16 additionally require that the voice traffic is analog voice traffic, and thereby introduce a patentably distinct element in addition to the elements recited in Claims 1, 8 and 15, respectively. The cited combination of Kato and Kim, however, does not teach or suggest that the voice traffic is analog voice traffic in combination with the base claim limitations. Thus, the combination of Kato and Kim does not render obvious dependent Claims 2, 9 and 16 as asserted by the Examiner. Accordingly, the Appellants respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 2, 9 and 16 and allow issuance thereof.

C. Rejection of Claims 4, 11 and 17

The Examiner has rejected Claims 4, 11 and 17 under 35 U.S.C. §103(a) as being unpatentable over Kato in view of Kim. The above argument establishing that the combination of Kato and Kim does not teach or suggest each element of the inventions of independent Claims 1, 8 and 15 is incorporated herein by reference. Dependent Claims 4, 11 and 17 additionally require that the data comprises caller identification data, and thereby introduce a patentably distinct element in addition to the elements recited in Claims 1, 8 and 15, respectively. The cited combination of Kato and Kim, however, does not teach or suggest that the data comprises caller identification data in combination with the base claim limitations. Thus, the combination of Kato and Kim does not render obvious dependent Claims 4, 11 and 17 as asserted by the Examiner. Accordingly, the Appellants respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 4, 11 and 17 and allow issuance thereof.

D. Rejection of Claims 5, 12 and 18

The Examiner has rejected Claims 5, 12 and 18 under 35 U.S.C. §103(a) as being unpatentable over Kato in view of Kim. The above argument establishing the combination of Kato and Kim does not teach or suggest each element of the inventions of independent Claims 1, 8 and 15 is incorporated herein by reference. Dependent Claims 5, 12 and 18 additionally require that the data comprises menu item selection data, and thereby introduce a patentably distinct element in addition to the elements recited in Claims 1, 8 and 15, respectively. The cited combination of Kato and Kim, however, does not teach or suggest that the data comprises menu item selection data in combination with the base claim limitations. Thus, the combination of Kato and Kim does not render obvious

dependent Claims 5, 12 and 18 as asserted by the Examiner. Accordingly, the Appellants respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 5, 12 and 18 and allow issuance thereof.

E. Rejection of Claims 6, 13 and 19

The Examiner has rejected Claims 6, 13 and 19 under 35 U.S.C. §103(a) as being obvious over Kato in view of Kim. The above argument establishing that the combination of Kato and Kim does not teach or suggest each element of the inventions of independent Claims 1, 8 and 15 is incorporated herein by reference. Dependent Claims 6, 13 and 19 additionally require that the voice traffic is transmitted in frames, and thereby introduce a patentably distinct element in addition to the elements recited in Claims 1, 8 and 15, respectively. The cited combination of Kato and Kim, however, does not teach that the voice traffic is transmitted in frames in combination with the base claim limitations. Thus, the combination of Kato and Kim does not render obvious dependent Claims 6, 13 and 19 as asserted by the Examiner. Accordingly, the Appellants respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 6, 13 and 19 and allow issuance thereof.

F. Rejection of Claims 21-22

The Examiner has rejected Claims 21-22 under 35 U.S.C. §103(a) for being obvious over Kato in view of Kim. The above argument establishing that the combination of Kato and Kim does not teach or suggest each and every element of independent Claims 1 and 8 is incorporated herein by reference. Dependent Claims 21-22 additionally require that the voice traffic and the data are

received from a coupled telephone line, and thereby introduce a patentably distinct element in addition to the elements recited in Claims 1 and 8, respectively. The cited combination of Kato and Kim, however, does not teach the voiee traffic and the data are received from a coupled telephone line in combination with the base claim limitations.

Additionally, Kato teaches against the voice traffic **and** the data are received from a coupled telephone line since Kato teaches the data is transmitted from a mobile packet data station to a base station and the voice traffic occurs between a mobile voice station and the base station. (*See* Figure 1.) Thus, for at least these reason, the cited combination of Kato and Kim does not provide a *prima facie* case of obvious of dependent Claims 21-22 as asserted by the Examiner. Accordingly, the Appellants respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 21-22 and allow issuance thereof.

Rejection under 35 U.S.C. 103(a) over Kato in view of Kim and in further view of Walley

A. Rejection of Claims 7, 14 and 20

The Examiner has rejected Claims 7, 14 and 20 under 35 U.S.C. §103(e) as being unpatentable over Kato in view of Kim and in further view of Walley. The above argument establishing that the combination of Kato and Kim does not teach or suggest each element of the inventions of independent Claims 1, 8 and 15 is incorporated herein by reference. Dependent Claims 7, 14 and 20 additionally require identifying a pause in voice traffic by comparing a peak energy of the voice traffic to a noise floor reference, and thereby introduce a patentably distinct element in addition to the elements recited in Claims 1, 8 and 15, respectively. As recognized by the Examiner, the combination of Kato and Kim does not teach or suggest identifying a pause in voicc traffic by

comparing a peak energy of the voice traffic to a noise floor reference. To cure this deficiency, the Examiner cites Walley.

Walley, therefore, has been cited to teach the subject matter of dependent Claims 7, 14 and 20 and not to cure the noted deficiency of the combination of Kato and Kim regarding independent Claims 1, 8 and 15. Additionally, the Appellants do not find where Walley teaches identifying a pause in voice traffic that is to be transmitted over the voice channel and responding to the pause by causing the transmitter to transmit the data to the receiver over the voice channel. On the contrary, Walley is directed to generating a signal quality value using a simple, robust method. (*See* column 2, lines 20-23.)

Thus, the cited combination of Kato, Kim and Walley does not provide a *prima facie* case of obviousness of dependent Claims 7, 14 and 20, which include the elements of the respective independent claims. Accordingly, Claims 7, 14 and 20 are nonobvious over the cited combination of Kato over Kim in view of Walley. The Appellants, therefore, respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of Claims 7, 14 and 20 and allow issuance thereof.

For the reasons set forth above, the Claims on appeal are patentably nonobvious over Kato in view of Kim and in further view of Walley. Accordingly, the Appellants respectfully request that the Board of Patent Appeals and Interferences reverse the Examiner's Final Rejection of all of the Appellant's pending claims.

Respectfully submitted,

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VIII. APPENDIX A - CLAIMS

1. For use in communicating data over a voice channel between a transmitter of a base station and a receiver of a handset of a cordless telephone, a system comprising:

a silence detector, coupled to said transmitter, that identifies a pause in voice traffic that is to be transmitted over said voice channel and generates an interjection signal during said pause; and

a data injector, coupled to said silence detector, that receives said interjection signal and responds by causing said transmitter to transmit data to said receiver over said voice channel.

2. The system as recited in Claim 1 wherein said voice traffic is analog voice traffic.

3. (canceled)

4. The system as recited in Claim 1 wherein said data comprises caller identification data.

5. The system as recited in Claim 1 wherein said data comprises menu item selection data.

6. The system as recited in Claim 1 wherein said transmitter transmits said voice traffic in frames.

7. The system as recited in Claim 1 wherein said silence detector identifies said pause by comparing a peak energy of said voice traffic to a noise floor reference.

8. A method of communicating data over a voice channel between a transmitter of a base station and a receiver of a handset of a cordless telephone, comprising:

identifying a pause in voice traffic that is to be transmitted over said voice channel; and

responding to said pause by causing said transmitter to transmit data to said receiver over said voice channel.

9. The method as recited in Claim 8 wherein said voice traffic is analog voice traffic.

10. (canceled)

11. The method as recited in Claim 8 wherein said data comprises caller identification data.

12. The method as recited in Claim 8 wherein said data comprises menu item selection data.

13. The method as recited in Claim 8 wherein said transmitter transmits said voice traffic in frames.

14. The method as recited in Claim 8 wherein said identifying comprises comparing a peak energy of said voice traffic to a noise floor reference.

15. A cordless telephone, comprising:

a base station transceiver;

a handset transceiver, said base station and handset transceivers cooperable to establish a voice channel therebetween;

a silence detector, coupled to said base station transceiver, that identifies a pause in voice traffic that is to be transmitted over said voice channel and generates an interjection signal during said pause; and

a data injector, coupled to said silence detector, that receives said interjection signal and responds by causing said base station transceiver to transmit data to said receiver over said voice channel.

16. The cordless telephone as recited in Claim 15 wherein said voice traffic is analog voice traffic.

17. The cordless telephone as recited in Claim 15 wherein said data comprises caller identification data.

18. The cordless telephone as recited in Claim 15 wherein said data comprises menu item selection data.

19. The cordless telephone as recited in Claim 15 wherein said base station transceiver transmits said voice traffic in frames.

20. The cordless telephone as recited in Claim 15 wherein said silence detector identifies said pause by comparing a peak energy of said voice traffic to a noise floor reference.

21. The system as rccited in Claim 1 wherein said system receives said voice traffic and said data from a tclcphone line coupled thereto.

22. The method as recited in Claim 8 further comprising receiving said voice traffic and said data from a telephone line coupled to said base station.

IX. APPENDIX B - EVIDENCE

The evidence in this appendix includes U.S. patents to Kato, Kim and Walley. Kim and Kato were entered in the record by the Examiner with the seventh Examiner's Office Action. Walley was entered in to the record by the Examiner with the fifth Examiner's Office Action.

US006044266A

United States Patent [19]

Kato

[11] **Patent Number: 6,044,266**

[45] **Date of Patent: Mar. 28, 2000**

[54] **METHOD AND APPARATUS FOR TRANSMITTING DATA PACKETS OVER VOICE CHANNEL**

[75] Inventor: **Hidenori Kato**, Tokyo, Japan

[73] Assignee: **NEC Corporation**, Tokyo, Japan

[21] Appl. No.: **08/916,268**

[22] Filed: **Aug. 22, 1997**

[30] **Foreign Application Priority Data**

Aug. 27, 1996 [JP] Japan 8-224772

[51] **Int. Cl.**^{7} **H04Q 7/20**

[52] **U.S. Cl.** **455/422**; 455/466; 455/455; 455/445; 455/450

[58] **Field of Search** 455/422, 432, 455/466, 456, 450, 434, 436, 438, 419, 445, 464; 370/433, 435, 465, 528

[56] **References Cited**

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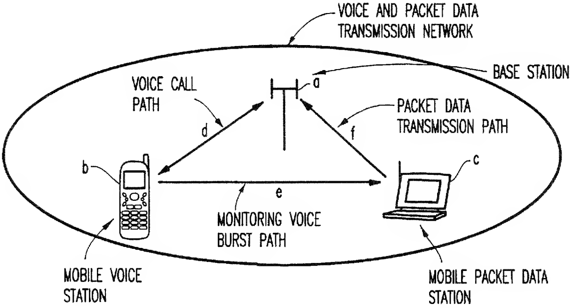
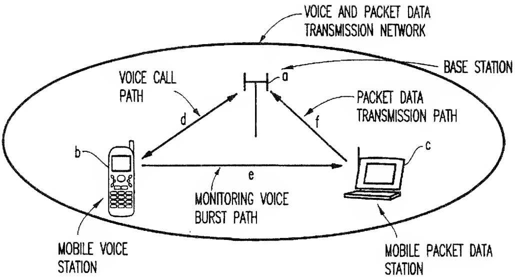
Primary Examiner—Daniel S. Hunter
Assistant Examiner—Yemane Woldetatios
Attorney, Agent, or Firm—McGinn & Gibb, P. C.

[57] **ABSTRACT**

A communication system includes a first station in voice communication with a base station and a second station in data communication with the base station. The second station monitors the voice communication and identifies predetermined periods (e.g., preferably silent periods) of the voice communication. The second station transmits data packets during the predetermined periods.

29 Claims, 9 Drawing Sheets

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PATH
d
PACKET DATA
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BURST PATH
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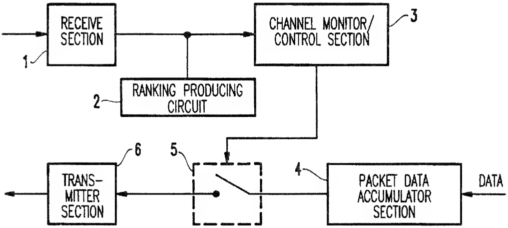
STATION
MOBILE PACKET DATA
STATION

FIG.1A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CONTROL SECTION
3
RANKING PRODUCING
CIRCUIT
2
TRANS-
MITTER
SECTION
6
5
4
PACKET DATA
ACCUMULATOR
SECTION
DATA

FIG.1B

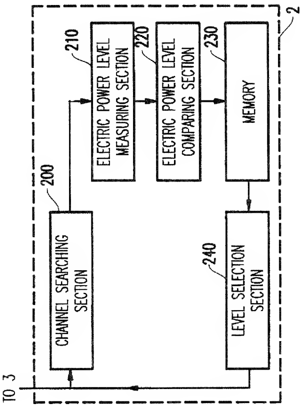


FIG.2A

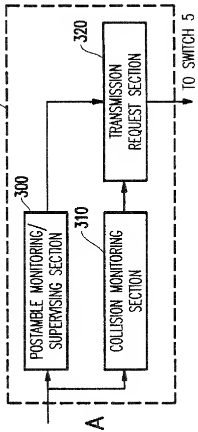
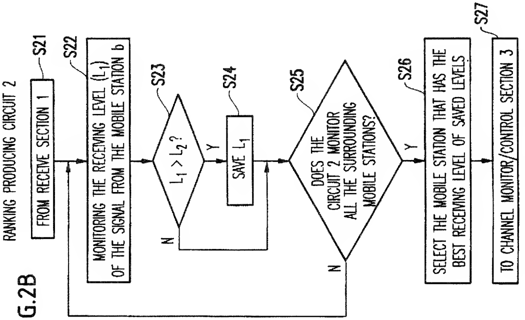


FIG.3A

FIG.2B



CHANNEL MONITOR/CONTROL SECTION 3

FIG.3B

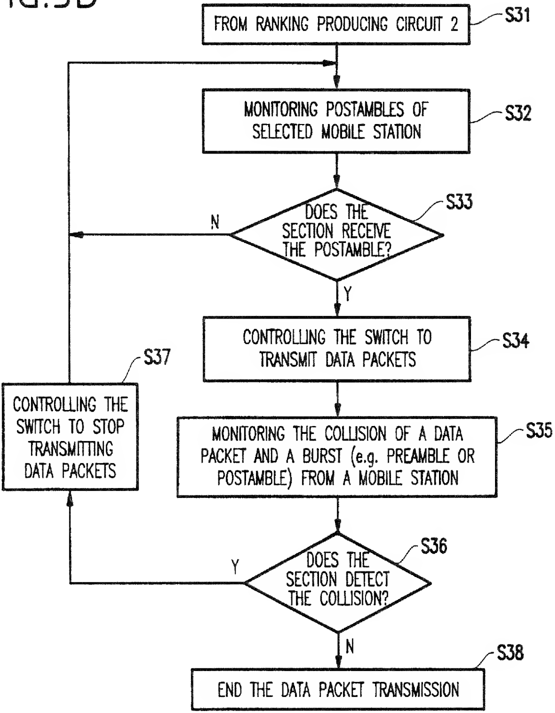
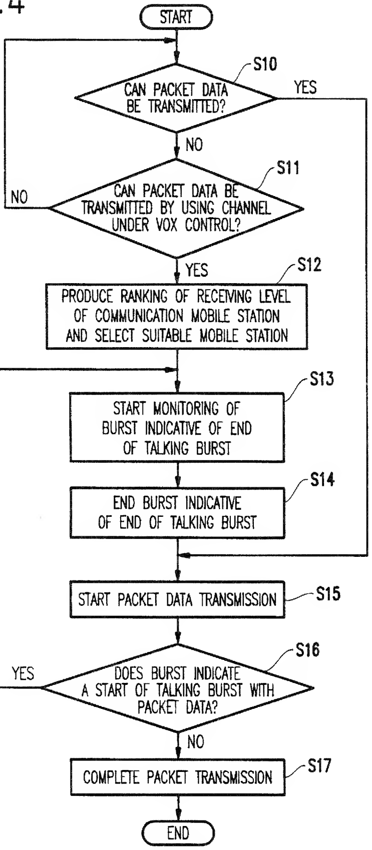


FIG.4


START
S10
CAN PACKET DATA BE TRANSMITTED?
YES
NO
S11
CAN PACKET DATA BE TRANSMITTED BY USING CHANNEL UNDER VOX CONTROL?
NO
YES
S12
PRODUCE RANKING OF RECEIVING LEVEL OF COMMUNICATION MOBILE STATION AND SELECT SUITABLE MOBILE STATION
S13
START MONITORING OF BURST INDICATIVE OF END OF TALKING BURST
S14
END BURST INDICATIVE OF END OF TALKING BURST
START PACKET DATA TRANSMISSION
S15
S16
DOES BURST INDICATE A START OF TALKING BURST WITH PACKET DATA?
YES
NO
COMPLETE PACKET TRANSMISSION
S17
END

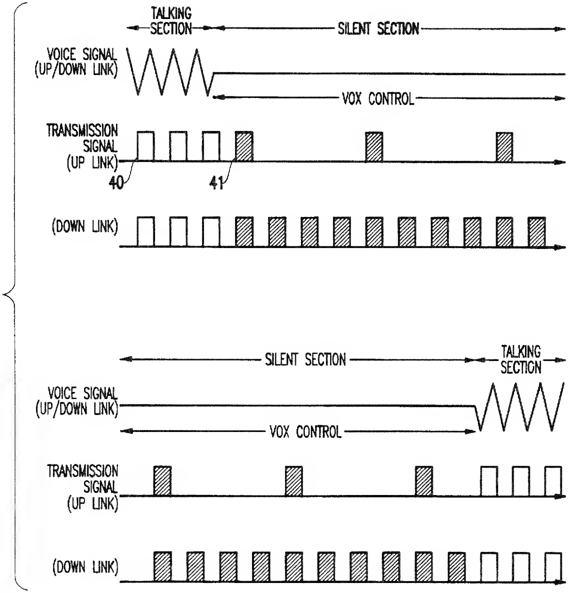

TALKING SECTION
SILENT SECTION
VOICE SIGNAL (UP/DOWN LINK)
VOX CONTROL
TRANSMISSION SIGNAL (UP LINK)
40
41
(DOWN LINK)
SILENT SECTION
TALKING SECTION
VOICE SIGNAL (UP/DOWN LINK)
VOX CONTROL
TRANSMISSION SIGNAL (UP LINK)
(DOWN LINK)

FIG.5

FIG.6A

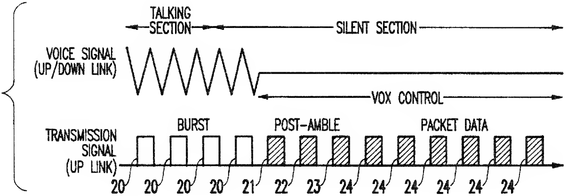
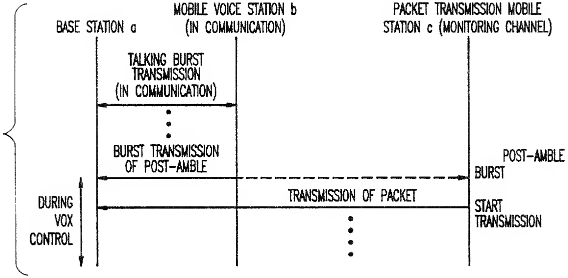

TALKING SECTION
SILENT SECTION
VOICE SIGNAL (UP/DOWN LINK)
VOX CONTROL
TRANSMISSION SIGNAL (UP LINK)
BURST
POST-AMBLE
PACKET DATA
20
20
20
20
21
22
23
24
24
24
24
24
24

FIG.6B


BASE STATION a
MOBILE VOICE STATION b (IN COMMUNICATION)
PACKET TRANSMISSION MOBILE STATION c (MONITORING CHANNEL)
TALKING BURST TRANSMISSION (IN COMMUNICATION)
BURST TRANSMISSION OF POST-AMBLE
POST-AMBLE BURST
TRANSMISSION OF PACKET
START TRANSMISSION
DURING VOX CONTROL

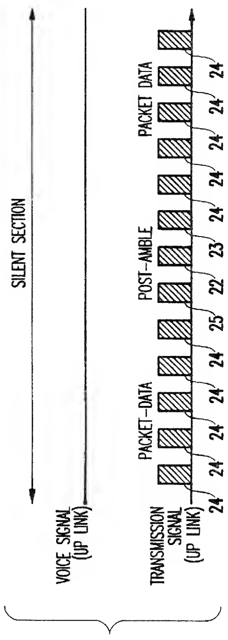

SILENT SECTION
VOICE SIGNAL (UP LINK)
TRANSMISSION SIGNAL (UP LINK)
PACKET-DATA
POST-AMBLE
PACKET DATA
24
25
22
23

FIG.7A

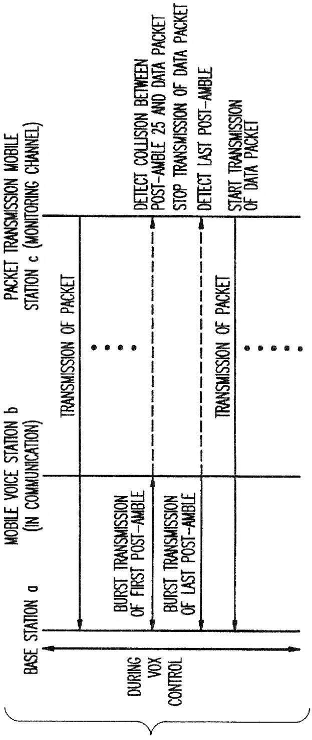

BASE STATION a
MOBILE VOICE STATION b (IN COMMUNICATION)
PACKET TRANSMISSION MOBILE STATION c (MONITORING CHANNEL)
TRANSMISSION OF PACKET
BURST TRANSMISSION OF FIRST POST-AMBLE
DETECT COLLISION BETWEEN POST-AMBLE 25 AND DATA PACKET
STOP TRANSMISSION OF DATA PACKET
BURST TRANSMISSION OF LAST POST-AMBLE
DETECT LAST POST-AMBLE
TRANSMISSION OF PACKET
START TRANSMISSION OF DATA PACKET
DURING VOX CONTROL

FIG.7B

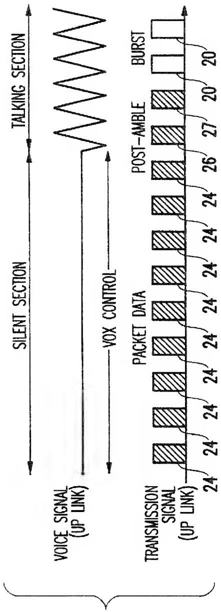

TALKING SECTION
SILENT SECTION
VOICE SIGNAL (UP LINK)
VOX CONTROL
TRANSMISSION SIGNAL (UP LINK)
PACKET DATA
POST-AMBLE
BURST
24
26
27
20

FIG.8A

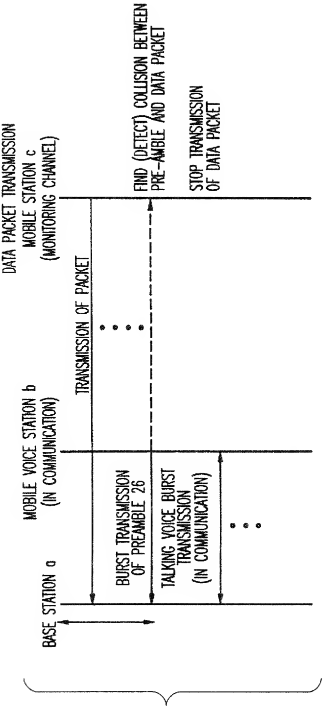

DATA PACKET TRANSMISSION MOBILE STATION c (MONITORING CHANNEL)
MOBILE VOICE STATION b (IN COMMUNICATION)
BASE STATION a
TRANSMISSION OF PACKET
FIND (DETECT) COLLISION BETWEEN PRE-AMBLE AND DATA PACKET
STOP TRANSMISSION OF DATA PACKET
BURST TRANSMISSION OF PREAMBLE 26
TALKING VOICE BURST TRANSMISSION (IN COMMUNICATION)

FIG.8B

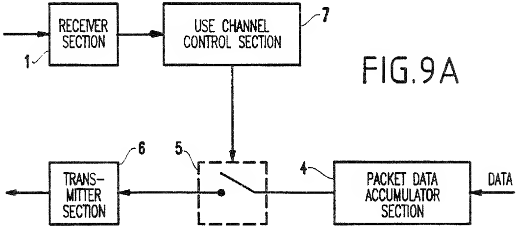

RECEIVER SECTION
1
USE CHANNEL CONTROL SECTION
7
TRANS-MITTER SECTION
6
5
4
PACKET DATA ACCUMULATOR SECTION
DATA

FIG.9A

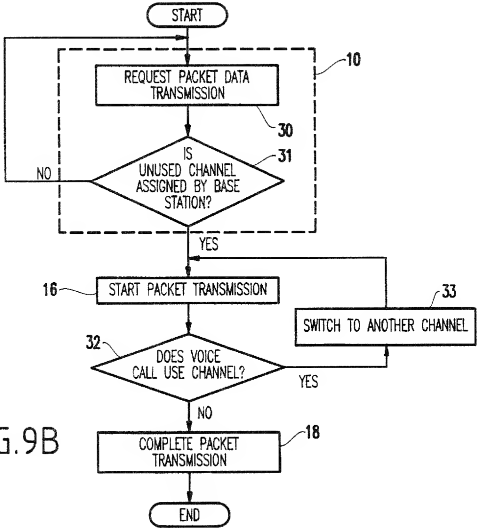

START
10
REQUEST PACKET DATA TRANSMISSION
30
31
IS UNUSED CHANNEL ASSIGNED BY BASE STATION?
NO
YES
16
START PACKET TRANSMISSION
33
SWITCH TO ANOTHER CHANNEL
32
DOES VOICE CALL USE CHANNEL?
YES
NO
COMPLETE PACKET TRANSMISSION
18
END

FIG.9B

METHOD AND APPARATUS FOR TRANSMITTING DATA PACKETS OVER VOICE CHANNEL

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention generally relates to a method and apparatus for acquiring a channel in radio packet-type data communication which is employed in mobile communications such as a cellular phone system. More particularly, the present invention relates to a method and apparatus for acquiring a channel in radio packet-type data communication, in which the transmission status of a mobile terminal using a given channel, is monitored such that, when the monitored terminal is silent and the channel is unused, packets of data are transmitted over the channel. The present invention makes communications systems more efficient by transmitting packet-type data during periods when channels are otherwise unused (e.g., so-called "silent periods").

2. Description of the Related Art

As transmission information becomes more complex and with the wider use of multimedia, information must be transferred more efficiently. Additionally, when radio data communication is performed between a mobile terminal and a base station, a packet-type communication system is used for effectively utilizing a limited number of radio channels so as to minimize communication cost and to serve more users.

For example, a typical packet-type communication system is a Cellular Digital Packet Data (CDPD) system. The CDPD system includes a CDPD base station, which can be used with a base station for a cellular phone system now in service. The CDPD base station identifies (and assigns) radio channels which are not used for voice communication, and transmits data packets over the unused channels. Thus, radio channels not being used for voice communication are selectively used for data communication.

In such a system, voice communication is given preference in utilizing the radio channels. Thus, if a request exists to use a channel for voice communication, the data communication concedes the channel to the voice communication. Hence, even if a channel is being used for data communication, upon a request for a mobile voice unit, the data communication is terminated, and must wait for another channel to become available (or "hops" to another channel if one is available).

FIG. 9(*a*) is a block diagram of a packet-type transmitter, and FIG. 9(*b*) is a block diagram illustrating the operation of the packet-type transmitter.

In operation, a use channel control section **7** controls which channels are used for data transmission based on channel state information (e.g., information regarding which channels are not being used for voice communication). A receiver section **1** indicates which channels are not being used. A switch **5** performs an ON-OFF switching operation, under the control of the channel control section **7**, to control whether voice information or packet-type data is transmitted over a given channel.

As shown in FIG. 9(*b*), in step **30** the operation begins with a request for a packet-type data transmission being made to the base station. In block **31**, a determination is made as to whether there is an unused channel assigned by the base station. In other words, it is determined whether an available channel (e.g., a channel not being used by another mobile unit) exists. If there is no available channel, the process returns to step **30** to again request a packet-type data transmission.

If a channel is available, in step **16**, packet transmission is started. In step **32**, it is determined whether a mobile voice station requests the channel being used for packet-type data transmission and more specifically whether a voice call currently uses the channel. If "YES" in step **32**, processing proceeds to step **33**, and a search is begun for a different available channel. If there is no voice request (e.g., a "NO" in step **32**), the processing proceeds to step **18**, where the packet-type data transmission is completed, and the process terminates.

In conventional packet-type data communication systems, voice communication is given preference, and a channel is used exclusively for voice or exclusively for data (e.g., conventional systems do not share voice and packet-type data on the same channel). Therefore, with conventional systems, if all of the available channels are transmitting voice communication, data cannot be transmitted. This is a problem and lowers system efficiency.

Hence, if a voice communication (for example, a cellular telephone call) is in progress, the channel used by the cellular phone cannot be used to transmit data packets during the entire length of the call (or during the time that the cellular phone remains within a given cell). Further, the channel is reserved for voice communication upon receiving a request for connection for voice communication. In conventional systems, voice communication dominates the channel availability, regardless of whether actual voice bursts are being transmitted and received. Therefore, the conventional system does not use the "dead air" time (e.g., the time period between words or other silent periods of voice communication, or the time after a request for voice channel connection is received but no actual voice communication is occurring) to transmit data or other signals.

Additionally, if a channel is being used for the packet-type data communication, it will be interrupted and terminated if there is a request for voice communication. The above-mentioned problems result in the conventional systems having poor efficiency.

JPA 2-117227 discloses the general concept of transmitting data packets over the voice channel. However, for data communication, JPA '227 uses/requires an idle (e.g., vacant) channel not being used for voice communication. It cannot transmit packet data during a silent period in a channel being used for voice communication.

SUMMARY OF THE INVENTION

In view of the foregoing and other problems of the conventional systems, it is an object of the present invention to simultaneously use a radio channel for voice communication and for packet-type data communication.

Another object of the present invention is to reduce congestion or interruption during packet-type data communication.

In a first aspect of the present invention, a system for acquiring a channel in radio packet data communication includes a receiver section for receiving a voice burst transmitted from a mobile station in communication with a base station and measuring receiving levels, a ranking producing section for producing a ranking list of the receiving state based on receiving levels from the output of the receiver section radio waves transmitted from the communicating mobile station, and a channel monitor/control section for selecting a channel with a predetermined receiving state from the output from the ranking producing section and monitoring the receiving state of the channel.

When the mobile station is determined to have a state for not sending a voice burst (e.g., no voice burst is being

transmitted), packet data is transmitted by interrupting the channel on which the voice burst is not transmitted for transmitting packet data.

When the terminal in voice communication transmits a voice burst again during transmission of packet data, the data communication is suspended, and it is determined whether another mobile terminal has a state for not transmitting a voice burst (e.g., no voice burst is being transmitted), thereby a channel for radio packet data communication is obtained by repeating such steps.

In another aspect of the present invention, a data communication method and system is provided for use with a base station, a mobile voice station in voice communication with the base station and a mobile data station in data communication with the base station. The mobile data station monitors the voice communication and identifies silent periods of the voice communication. The silent periods include periods between sounds. The mobile data station transmits data packets during the silent periods.

Unlike the conventional systems such as JPA '227, the present invention does not require an idle (vacant) channel, but can transmit data during a silent period in a channel being used for voice communication. Thus, overall system efficiency is increased, and congestion and interruption during packet-type data communication is reduced.

BRIEF DESCRIPTION OF THE DRAWINGS

The foregoing and other objects, aspects and advantages will be better understood from the following detailed description of a preferred embodiment of the invention with reference to the drawings, in which:

FIG. **1**(*a*) is a schematic diagram of a packet-type transmission system according to an embodiment of the present invention;

FIG. **1**(*b*) is a schematic block diagram of a packet-type transmitter of the system according to the present invention;

FIG. **2**(*a*) illustrates the structure of a ranking producing circuit **2** of the packet-type transmitter of FIG. **1**(*b*);

FIG. **2**(*b*) illustrates a flowchart of the operations of the ranking producing circuit **2**;

FIG. **3**(*a*) illustrates the structure of a channel monitor/control section **3** of the packet-type transmitter of FIG. **1**(*b*);

FIG. **3**(*b*) illustrates a flowchart of the operations of the channel monitor/control section **3**;

FIG. **4** is a flowchart illustrating the operation of the embodiment of the inventive packet-type transmission system;

FIG. **5** illustrates schematic wave-form diagrams of voice and data signals according to the present invention;

FIG. **6**(*a*) is a schematic wave-form diagram illustrating an ending voice burst of the present invention;

FIG. **6**(*b*) is a schematic timing diagram illustrating the operation of the present invention during an ending voice burst;

FIG. **7**(*a*) is a schematic wave-form diagram illustrating a silent period of the present invention;

FIG. **7**(*b*) is a schematic timing diagram illustrating the operation of the present invention during a silent period;

FIG. **8**(*a*) is a schematic wave-form diagram illustrating a beginning voice burst of the present invention;

FIG. **8**(*b*) is a schematic timing diagram illustrating the operation of the present invention during a beginning voice burst;

FIG. **9**(*a*) is a schematic block diagram of a conventional packet-type transmitter; and

FIG. **9**(*b*) is a flowchart illustrating the operation of the conventional packet-type transmission system of FIG. **9**(*a*).

DETAILED DESCRIPTION OF A PREFERRED EMBODIMENT OF THE INVENTION

Referring now to the drawings, and more particularly to FIG. **1**(*a*), a packet-type transmission system according to the present invention is illustrated. In FIG. **1**(*a*), a is a base station, b is a mobile station for voice communication (e.g., a "mobile voice station"), c is a mobile station for packet transmission (e.g., a "mobile data station"), d is a call path for the mobile voice station b, e is a path for monitoring voice bursts (e.g., signals containing voice transmissions) from the mobile voice station b, and f is a packet transmission path for the mobile data station c.

The base station a engages in voice message communication with the mobile voice station b through the call path d, and the mobile data station c monitors the state of voice bursts being transmitted from the mobile voice station b through the burst monitor path e. The mobile data station c transmits data to the base station a through the packet transmission path f only when the voice burst monitor path e indicates that the call path d is silent (e.g., no voice burst is being transmitted). Typically, the mobile data station c polls all stations b and selects the best channel from them, as described below.

FIG. **1**(*b*) is a block diagram of a packet-type transmitter according to an embodiment of the present invention located within, for example, mobile data station c. A receiver section **1** monitors a voice burst being transmitted from various mobile voice stations b to base station a, and measures the signal level of the voice bursts. A ranking producing section **2** continually measures a level of electrical power, and also produces a ranking list of the different signal levels output from the receiver section **1** according to levels of the radio waves transmitted by the mobile voice stations b.

A channel monitor/control section **3** receives a channel number level and the input signal, selects a channel with the best receiving state from the output of the ranking producing circuit **2**, and monitors the receiving state of that channel. The best receiving state is determined as the strongest signal. Specifically, it is based on the channel which has the strongest signal because the VOX control of such a channel is monitored most correctly (e.g., precisely).

Each of the mobile voice stations b transmits an ending voice burst (indicating the end of a voice burst) which will be followed by a certain period of silence (e.g., during which the user is not talking on a channel (e.g., an uplink), but he may be listening to the other party (e.g., on the downlink)). The ending voice burst is discussed in greater detail below with regard to FIG. **6**(*a*) The mobile stations b also transmit a starting voice burst indicating the starting of a series of voice bursts. The starting voice burst is discussed in greater detail below with regard to FIG. **8**(*a*). Between the ending voice burst and the next subsequent (beginning) voice burst is silence. The silent state is discussed below with reference to FIG. **7**(*a*).

This series of ending and beginning voice bursts is called Voice-Operated Transmission (VOX). The VOX control for mobile voice station b can be divided into three patterns. The first pattern occurs when the mobile voice station b changes from the talking state to the silent state (e.g., an ending voice burst), the second pattern occurs when the mobile voice station b remains unchanged in the silent state, and the third pattern occurs when the mobile voice station b changes from the silent state to the talking state (e.g., a beginning voice burst).

The channel monitor/control section **3** monitors VOX control information of the mobile station b and the receiving level at the terminal c. The section **3** monitors the receiving level of the signal from the mobile voice station b. The monitor/control section **3** controls a switch **5**, which is movable between first and second positions (e.g., an active position and an inactive position). The switch is activated (turned ON) when the monitor/control section **3** determines that a radio channel can be used, and is inactivated (turned OFF) when the monitor/control section **3** determines that the radio channel cannot be used. The monitor/control section **3** constantly monitors the various channels available to a respective base station. When an ending voice burst is detected, transmission of packets of data is commenced on that channel by activating the switch **5**, and the packets of data continue to be transmitted until a beginning voice burst signal is received.

A packet-type data accumulator section **4** accumulates data to be transmitted, in the form of data packets. The switch **5** starts and stops the transmission of the data packets according to information from the channel monitor/control section **3**.

A transmitter section **6** performs transmission over the channel selected by the channel monitor/control section **3**.

Referring now to FIG. **2**(*a*), the structure of the ranking producing circuit is shown in detail. As shown in the FIG. **2**(*a*), the ranking producing circuit includes a channel searching section **200**, an electric power level measuring section **210**, an electric power level comparing section **220**, a memory **230**, and a level selection section **240**.

First, the channel searching section **200** is for searching/monitoring all of the surrounding mobile voice stations b based on inputs from the ranking producing circuit **2**. Specifically, the channel searching section **200** receives an input signal from receiver section **1**, and provides an output (relating to a channel) to the electric power level measuring section **210**. As a result, section **210** monitors and measures the receiving level (L1) of the signal from a mobile voice station b.

Based on an output from section **210**, the electric power level comparing section **220** compares the electric power level L1 with a predetermined value (e.g., a predetermined threshold level L2), to determine whether L1 is greater than the predetermined value.

If L1 is greater than the predetermined value L2, then level L1 is saved (along with the channel number associated with the level L1) in memory **230**. Memory **230** stores the channel number level of each channel above the threshold in a table or the like. If the determination is "NO" (e.g., L1 is less than L2), then the channel number and level are not stored.

Based on an output from the memory **230**, a level selection section **240** selects the channel with the highest level. That is, the mobile voice station b is selected that has the best receiving level of the saved levels.

Thereafter, the level selection section **240** provides an output to the channel monitor/control section **3** representing the mobile station b having the best receiving level of the saved levels.

Turning to FIG. **2**(*b*), the operation of the ranking producing circuit **2** will be described in further detail. First, in step **S21**, a signal is received by the channel searching section **200** from the receiver section **1**. Thereafter, in step **S22**, the receiving level (**L1**) of the signal from the mobile voice station b is monitored/measured by the electric power level measuring section **210** based on an output from the channel searching section **200**.

In step **S23**, it determined by the electric power level comparing section **220** whether L1 is greater than a predetermined value (e.g., a predetermined level L2).

If the determination is "YES" in step **S23**, then the process proceeds to step **S24** and L1 is saved in memory **230**, and the process proceeds to step **S25**.

Conversely, if the determination in step S23 is "NO" (e.g., L1 is less than L2), then the process immediately proceeds to step **S25**.

In step **S25**, it is determined whether all of the surrounding mobile voice stations b are (e.g., have been) monitored by ranking producing circuit **2**. If the determination is "NO", the process loops back to step **S22** . If the determination is "YES", then the process continues to step **S26**.

In step **S26**, the level selection section **240** selects the mobile voice station b that has the best receiving level of the saved levels in the memory **230**.

Thereafter, in step **S27**, an output is provided to the channel monitor/control section **3** representing the mobile station b having the best receiving level of the saved levels.

Turning now to FIG. **3**(*a*), the structure of the channel monitor/control section **3** will be described in further detail. The channel monitor/control section **3** includes a postamble monitoring/supervising section **300**, a collision monitoring section **310**, and a transmission request section **320**.

Specifically, the postamble monitoring/supervising section **300** receives an output from the ranking producing circuit **2** representing the mobile voice station b having the best receiving levels of the saved levels, and monitors the postambles of the selected mobile voice station b (e.g., determines whether the postambles have been received). Thus, section **300** determines whether it has received the postambles of the selected mobile voice station b. If the determination is "YES", then the postamble monitoring/supervising section **300** provides an output to the transmission request section **320**, and the switch **5** is controlled to transmit the data packets. If section **300** determines that it has not received the postambles, then the section **300** simply continues to wait for a postamble, and the switch **5** is unchanged from its current mode.

The collision monitoring section **310** also receives the output from the ranking producing circuit **2**. Section **310** detects whether an actual collision has occurred between the voice transmission and the data packet transmission. Specifically, the collision of a data packet and a burst (e.g., a preamble or a postamble) from the mobile voice station b is monitored and determined by the collision monitoring section **310**. It is noted that section **310** simply discriminates the collision, as opposed to necessarily discriminating the object of the collision (e.g., preamble or postamble).

If the collision monitoring section **310** detects a collision, then the collision monitoring section **310** provides an output to the transmission request section **320**, thereby to control the switch **5** to stop transmitting the data packets and avoid further collision. By the same token, if the collision monitoring section **310** does not detect a collision, then the packet data transmission continues until it is completed.

Turning to FIG. **3**(*b*), the operation of the channel monitor/control section **3** will be described in further detail.

First in step **S31**, an output is received from the ranking producing circuit **2**.

Thereafter, in step **S32**, postambles of the selected mobile voice station b are monitored by the postamble monitoring/supervising section **300**.

In step **S33**, it is determined whether the section **3** has received the postamble. If the determination is "NO", then the process loops back to step **S32**.

If the determination is "YES", then the process continues to step S34 in which the switch 5 is controlled to transmit the data packets.

In step S35, the collision of a data packet and a burst (e.g., a preamble or a postamble) from the mobile voice station b is monitored. Thereafter, in step S36, it is determined whether the collision monitoring section 300 detects the collision.

If the collision monitoring section 300 detects the collision, the process flows to step S37, thereby to control the switch 5 (via an output from the collision monitoring section 310 to the transmission request section 320 to the switch 5) to stop transmitting the data packets.

If, in step S37, no collision is detected by the collision monitoring section 310, then the process continues to step S38, and the packet data transmission continues until it is completed.

Operation of the Invention

Hereinbelow, the operation of the foregoing embodiment of the present invention is described referring to FIG. 4.

First, the mobile data station c transmits a request for packet-type data transmission to the base station a, and it is determined whether the packet data can be transmitted (step S10). If "YES" in step S10 (e.g., if there is an unused channel) the packet transmission is started (step S15). An unused channel is one that is not being used currently for any type of voice or data transmission. Detection of such a channel is well-known in the art, and thus for brevity will not be described in detail herein.

If all channels are in use (e.g., a "NO" in step S10), the mobile data station c transmits a packet transmission request using a channel that is under VOX control (step S11). The mobile data station c determines that the channel is being used according to VOX control, based on an input from mobile voice station b. The channel monitor/control section 3 selects a channel which provides a good receiving state (step S12). Specifically, based on an input from the ranking producing circuit 2 which produces a ranking of the receiving level of the communicating mobile station in the manner described above, the channel monitor/control section 3 monitors the same and selects a suitable mobile station (e.g., the mobile station having the highest receiving state).

In step S13, the monitor/control section 3 monitors the selected channel for an ending voice burst. The ending voice burst is discussed below with respect to FIGS. 6(*a*) and 6(*b*).

In step S14, the ending voice burst is detected which indicates the end of talking. Accordingly , the channel is judged to be silent and not to include voice signals (until the beginning voice burst is detected). Hence, in step S15, the data packet transmission is started, and there will be no collision of voice and data during transmission.

During the packet-type data transmission, the control/monitor section 3 monitors the selected signal channel for a beginning voice burst (step S16), and specifically it is judged whether the burst indicates a start of talking (voice) burst with packet data. Detection of the beginning voice burst is well-known to persons ordinarily skilled in the art, and is also described in greater detail below with respect to FIGS. 8(*a*) and 8(*b*).

When the beginning voice burst is detected (e.g., a "YES" in step S16), the process loops again to step S13, and the packet-type data transmission is terminated to allow the voice signal to be transmitted from the mobile voice station b to the base station a. Alternatively, if "NO" is judged in step S16, the process continues to step S17, and the data packet transmission is completed.

It is noted that, prior to termination of the packet transmission, a state is entered for monitoring a burst indicating the end of VOX state and the beginning of the voice communication ("talking") state. If a collision between packet data and voice data occurs, the packet data transmission is stopped (as described above), and monitoring begins again for a burst indicating the end of a talking burst. A burst indicating the end of VOX state and the start of a talking burst would not interfere with the communicating mobile voice station b, as discussed below.

Specifically, even though a beginning voice burst and a data packet may collide, such a collision would not interfere with the communication of the mobile voice station b because even if the beginning voice burst collides and is lost by the action of bit interleave, transmission of the same information is contained in following (subsequent) voice bursts. Thus, for example, in FIG. 8(*a*), if preamble signal 26 is lost, the next preamble signal 27 still can be received. Specifically, the bast station a always monitors preamble signals from the mobile voice station b during the VOX control. Even if the preamble signal 26 is lost by the collision, the base station a still can receive the next preamble signal 27.

As mentioned above, when all packets of data have been transmitted by repeating a series of operations from steps S13 through S16, a request is made for terminating the packet transmission (step S17). Once all the packets of data have been successfully transmitted, the data packets are placed in proper (sequential) order and reassembled by a device connected to the base station a, as is well-known to those ordinarily skilled in the art.

Thus, with the invention, a radio channel can be simultaneously used for voice communication and for packet-type data communication, thereby increasing the system efficiency. Additionally, congestion or interruption during packet-type data communication is greatly reduced by the present invention, as compared to the conventional systems described above.

Second Embodiment

A second embodiment of the present invention is described with reference to FIG. 5. The second embodiment can be used with a base station that does not currently perform VOX control, because the second embodiment adjusts the base station so that it will perform VOX control. Thus, down-link packet data communication can be performed by using a channel under the VOX control by performing VOX control at the base station a, as well as at the mobile voice station b. The mobile data station c transmits packet data during the silent period. The present invention requires a channel having silent periods as a channel under VOX control. Thus, a channel having a silent period is found and used by the present invention.

Additionally, in cordless telephone systems, while VOX control is performed on both the mobile station and the base station, the VOX control for the cordless telephone systems is different from the VOX control for cellular telephone systems. Briefly, the differences in the VOX control of the two systems is that the VOX control in the cordless telephone system (e.g., such as a "personal handy system" (PHS) or a personal cellular system) transmits a VOX voice burst in every fourth frame during the VOX control, but does not have preambles and postambles.

The second embodiment of the present invention accommodates the differences between the different types of VOX controls by essentially ignoring the voice bursts during the silent periods. In the cordless telephone system, the first voice burst is lost by collision with a data packet, because the cordless telephone system does not have preambles. However, the lost burst can be ignored since its frame length is only 5 ms, and thus does not contain much information.

FIG. 5 illustrates wave-form transmission signals of a typical cordless telephone under VOX control. A down-link transmission signal transmits a VOX voice burst **41** during which packets of data cannot be transmitted. An up-link transmission signal also transmits a VOX voice burst **41** in every four frames during the VOX control. Briefly, the base station a transmits a burst during the silent period in the down-link VOX control because the VOX control aims to reduce electrical power consumed by the mobile voice station b. Thus, in the second embodiment of the present invention, the down-link VOX control is not accommodated because of not having any silent periods.

Therefore, with the present invention, data packets may be transmitted in the empty frames between the voice bursts **41**. However, as mentioned above, unlike cellular telephone systems, cordless telephones do not have beginning or ending voice bursts (e.g., there is no preamble/postamble at the end of VOX control), and a talking burst **40** is directly transmitted. In the second embodiment of the present invention, since the cordless telephone system does not have preambles, it stops transmitting packet data when packet data collides with a voice burst. Thus, the second embodiment of the present invention must simply wait until there is a collision, and then terminate a data packet transmission.

Further, with the second embodiment of the invention, since the talking voice burst **40** is directly transmitted, burst **40** may collide with a packet-type data burst. However, this loss does not cause significant problems because the frame length of the cordless telephone system is much shorter (e.g., as short as 5 ms) than the frame length for a cellular phone system (e.g., typically 20 ms). This loss is relatively unimportant because the amount of voice lost is typically undetectable by the user, and the loss is simply much less than the frame for a cellular telephone system. The lost data packet is retransmitted during the next silent period (e.g., when a VOX burst is detected). Thus, the control according to the present invention can be performed even on the cordless (e.g., a so-called personal cellular system or the like) telephone system.

FIGS. **6**(*a*)–**8**(*b*) illustrate a VOX operation, in which FIGS. **6**(*a*) and **6**(*b*) illustrate the operation of the invention during an ending voice burst, FIGS. **7**(*a*) and **7**(*b*) illustrate the operation of the invention during a silent period, and FIGS. **8**(*a*) and **8**(*b*) illustrate the operation of the invention during a beginning voice burst. FIG. **6**(*a*) illustrates an ending voice burst, and FIG. **6**(*b*) illustrates the packet transmission procedure during the ending voice burst. The mobile voice station b transmits a talking voice burst **20** when transmitting sound (e.g., a talking state) and an ending voice burst (e.g., post-amble) to indicate that a silent period will follow. The post-amble typically includes 448 bits (224 bits of unique word (PST**0**), and 224 bits of background noise generating information (PST1). However, since audio coding is interleaved for two slots, the post-amble is a signal across three bursts (e.g., these are not considered "voice bursts", but are standard for cellular systems such as a personal digital cellular system). The three bursts are (voice+PST**0**), (PST**0**+PST1), and (PST**1**+dummy), which correspond to reference numbers **21**, **22** and **23** in FIG. **6**(*a*).

As explained above, since the ending voice burst will be followed by silence, the mobile data station c transmits packets of data **24** during the silent portion (as illustrated in FIG. **6**(*a*)). VOX control is performed during the silent section of the voice signal (the uplink) and the sending of the transmission signal (including the transmission of the post-amble and the packet data).

FIGS. **7**(*a*) and **7**(*b*) illustrate a second example. FIG. **7**(*a*) is a wave-form pattern of the transmission signal under VOX control which indicates a silent voice burst (e.g., "no-voice" bursts). FIG. **7**(*b*) illustrates the packet transmission procedure during the silent state. The mobile voice station b periodically transmits post-ambles (e.g., no more than one per second), even if the silent state continues. Post-ambles are transmitted to let the base station know that the mobile voice station is still in a communication state.

An ending voice burst (e.g., the current beginning/first post-amble) is shown as burst **25** in FIG. **7**(*a*), which includes a dummy+PST**0**, instead of the ending voice burst **21** (voice+PST**0**) discussed with reference to FIG. **6**(*a*).

Then, such a silent post-amble collides with packets of data being transmitted by mobile data station c. Therefore, the mobile data station c stops the transmission of data packets, again enters into the channel monitoring state (e.g., monitoring for collisions of data packets), and resumes the transmission of data packet bursts when it finds a post-amble (**22**, **23**) from the channel of the communicating mobile voice station b. The last data packet is retransmitted once post-amble **23** is received.

The mobile data station c stops transmitting data packets when they collide with burst **25**, and starts retransmitting after receiving bursts **22** and **23**.

FIG. **8**(*a*) illustrates a beginning voice burst in which the voice signal of the communicating mobile station b changes from the silent state to the talking state, and the VOX control is terminated. FIG. **8**(*b*) illustrates a packet transmission procedure during the beginning voice burst.

When the communicating mobile voice station b changes from the silent state to the talking state, the mobile station b first transmits a preamble (PRE). Since audio coding is interleaved for two slots, the preamble becomes a signal across two bursts (e.g., these are not voice bursts). The two bursts are (dummy+PRE) and (PRE+voice), which correspond to reference numerals **26** and **27** of FIG. **8**(*a*).

When the top preamble burst **26** collides with a packet data burst **24** (e.g., is received by) the mobile data station c, the mobile data station c stops the transmission of the data packet burst, and again enters into the channel monitoring state to wait for the next ending voice burst (e.g., a post-amble).

With the unique and unobvious structure and method of the present invention, periods when channels, which are occupied by voice transmission, are silent, can be used advantageously. During the silent periods, the invention transmits data packets. Thus, the present invention can transmit data packets even when all available channels are engaged in voice communication. As a result, the present invention reduces congestion in radio communication channels and allows more data packets to be transferred than in the conventional systems.

Moreover, according to the present invention, since a channel is not occupied in transmitting data by performing VOX control, which stops the transmission when the silent state is attained during audio communication by another mobile station, and transmitting packet data over a channel on which a burst under the VOX control is not transmitted, if there is no empty channel when the packet data transmission is requested, the packet data communication can be performed by utilizing a gap in the voice communication under the VOX control, so that a channel is not occupied and can be efficiently utilized.

Additionally, according to the present invention, even if all channels are used, the packet data communication can be continued by transmitting packet data over a channel of another mobile station on which no burst under the VOX control is transmitted. Thus, unlike the conventional systems

and methods described above, congestion or interruption of packet data communication, which may be caused if voice communication uses all channels since it is given preference, may be reduced.

While the invention has been described with reference to the specific embodiments described above, it is not limited thereto and includes all variations which would be known to those ordinarily skilled in the art.

Having thus described my invention, what I claim as new and desire to secure by Letters Patent is as follows:

1. A system for acquiring a channel in radio packet data communication for use with a base station and a mobile station, in communication with said base station, for transmitting a voice burst, comprising:

a receiver for receiving said voice burst and measuring a receiving level thereof;

a ranking section for ranking a receiving state of said receiver based on receiving levels of radio waves transmitted from said mobile station; and

a channel monitoring and control section for selecting a channel with a predetermined receiving state based on an output from said ranking section, and for monitoring the receiving state of the channel,

wherein, for packet data communication, said mobile station selectively uses an idle channel and other channels engaged for voice communication by other mobile stations such that when said mobile station is determined not to be transmitting a voice burst on the channel, said channel is interrupted and packet data is transmitted thereover, and

wherein, when a voice burst is transmitted on said channel during transmission of said packet data, the transmission of packet data is suspended, and a second mobile station not transmitting a voice burst is identified, thereby to obtain said idle channel for said radio packet data communication.

2. The system according to claim 1, wherein, when said idle channel selected for the packet data transmission by the channel monitoring and control section is used for voice communication before termination of packet data transmission, said channel monitoring and controlling section suspends the transmission of packet data, monitors for a burst which indicates termination of the voice burst, and interrupts a channel not transmitting a voice burst.

3. The system according to claim 1, further comprising a switching circuit coupled to said channel monitoring and control section.

4. The system according to claim 3, wherein, when said channel monitoring and control section receives from said ranking section information indicating termination of the voice burst as a silent state, the channel monitoring and control section determines that the channel is usable, and activates said switching circuit to transmit packet data.

5. The system according to claim 4, wherein, when said channel monitoring and control section receives from said ranking section information indicating starting of the voice burst as a taking state, said channel monitoring and control section confirms, based on said information, a state where no burst is transmitted on the channel, determines that said channel cannot be used, and controls the packet data transmission to de-activate said switching circuit.

6. The system according to claim 2, further comprising a switching circuit coupled to said channel monitoring and control section.

7. The system according to claim 6, wherein, when said channel monitoring and control section receives from said ranking section information indicating termination of the voice burst as a silent state, the channel monitoring and control section determines that the channel is usable, and activates said switching circuit to transmit packet data.

8. The system according to claim 7, wherein, when said channel monitoring and control section receives from said ranking section information indicating starting of the voice burst as a talking state, said channel monitoring and control section confirms, based on said information, a state where no burst is transmitted on the channel, determines that said channel cannot be used, and controls the packet data transmission to de-activate said switching circuit.

9. The system according to claim 1, wherein said mobile station monitors channels used by said other mobile stations and selects one of a predetermined good quality channel for said packet data communication, and

wherein said channel monitoring and control section utilizes a VOX control to detect a silent period in an engaged channel for one of transmitting packet data and inserting packet data into said silent period.

10. The system according to claim 1, wherein said system is selectively employed in any one of said base station and said mobile station.

11. A system for acquiring a channel in radio packet data communication for use with a base station and a mobile station, in communication with said base station, for transmitting a voice burst, comprising:

a receiver for receiving said voice burst and measuring a receiving level thereof;

a ranking section for ranking a receiving state of said receiver based on receiving levels of radio waves transmitted from said mobile station; and

a channel monitoring and control section for selecting a channel with a predetermined receiving state based on an output from said ranking section, and for monitoring the receiving state of the channel,

wherein, when said mobile station is determined not to be transmitting a voice burst on the channel, said channel is interrupted and packet data is transmitted thereover, and

wherein, when a voice burst is transmitted on said channel during transmission of said packet data, the transmission of packet data is suspended, and a second mobile station not transmitting a voice burst is identified, thereby to obtain an idle channel for radio packet data communication,

wherein said ranking section comprises:

a power level measuring section for measuring a power level of said receiving levels of radio waves transmitted from said mobile station;

a power level comparing section for comparing said power level with a predetermined level;

a memory for storing levels of channels determined to have a level above said predetermined level and a respective channel number; and

a level selection section for selecting a channel having a highest level of said channels determined to have a level above said predetermined level, said level selection section providing an output to said channel monitoring and control section.

12. The system according to claim 11, wherein said channel monitoring and control section comprises:

a postamble monitoring/supervising section for receiving an output from said level selection section and for monitoring postambles of a selected channel, such that packet data is transmitted when a postamble has been received by said channel monitoring and control section; and

a collision monitoring section for determining whether a collision has occurred between said voice burst and said packet data,

wherein, when said collision monitoring section detects a collision, said collision monitoring section provides an output to terminate said radio packet data communication, and when no collision is detected by said collision monitoring section, said radio data packet communication continues to completion.

13. A system for acquiring a channel in radio packet data communication for use with a base station and a mobile station, in communication with said base station, for transmitting a voice burst, comprising:

a receiver for receiving said voice burst and measuring a receiving level thereof;

a ranking section for ranking a receiving state of said receiver based on receiving levels of radio waves transmitted from said mobile station; and

a channel monitoring and control section for selecting a channel with a predetermined receiving state based on an output from said ranking section, and for monitoring the receiving state of the channel,

wherein, when said mobile station is determined not to be transmitting a voice burst on the channel, said channel is interrupted and packet data is transmitted thereover, and

wherein, when a voice burst is transmitted on said channel during transmission of said packet data, the transmission of packet data is suspended, and a second mobile station not transmitting a voice burst is identified, thereby to obtain an idle channel for radio packet data communication,

wherein said channel monitoring and control section comprises:

a postamble monitoring/supervising section for receiving an output from said ranking section and for monitoring postambles of a selected channel, such that packet data is transmitted when a postamble is received by said channel monitoring and control section; and

a collision monitoring section for determining whether a collision has occurred between said voice burst and said packet data,

wherein, when said collision monitoring section detects a collision, said collision monitoring section provides an output to terminate said radio packet data communication, and when no collision is detected by said collision monitoring section, said radio data packet communication continues to completion.

14. A communication system comprising:

a base station;

a first station in voice communication with said base station; and

a second station in data communication with said base station,

wherein said second station monitors said voice communication and identifies a predetermined period of said voice communication, said second station transmitting data packets during said predetermined period,

wherein, for packet data communication, said second station selectively uses an idle channel and other channels engaged, for voice communication, by mobile stations.

15. The system according to claim **14**, wherein said first station comprises a first mobile station and said second station comprises a second mobile station, and

wherein said predetermined period comprises a silent period between audible sounds.

16. The system according to claim **15**, wherein said silent period comprises a period between an ending voice burst and a beginning voice burst.

17. The system according to claim **14**, wherein said voice communication and said data communication are made on a same channel.

18. The system according to claim **15**, further comprising a plurality of first mobile stations, and said base station communicates with said first mobile stations on separate channels,

wherein said second mobile station includes means for monitoring said channels and for transmitting said data packets on selected ones of said separate channels which include said predetermined period.

19. The system according to claim **14**, wherein said first station comprises a first mobile station and said second station comprises a second mobile station,

wherein said second mobile station monitors channels used by said first mobile station and selects one of a predetermined good quality channel for transmitting data packets in said packet data communication, and

wherein said second mobile station utilizes VOX control to detect a silent period in an engaged channel of said first mobile station for one of said transmitting of said data packets and inserting said data packets into said silent period.

20. A communication system comprising:

a base station:

a first station in voice communication with said base station:

a second station in data communication with said base station, wherein said second station monitors said voice communication and identifies a predetermined period of said voice communication, said second station transmitting data packets during said predetermined period, and wherein said first station comprises a first mobile station and said second station comprises a second mobile station, said predetermined period comprising a silent period between audible sounds; and

a plurality of first mobile stations, and said base station communicates with said first mobile stations on separate channels,

wherein said second mobile station includes means for monitoring said channels and for transmitting said data packets on selected ones of said separate channels which include said predetermined period, and means for ranking a receiving level of each of said channels.

21. The system according to claim **20**, wherein said means for ranking comprises:

a power level measuring section for measuring a power level of said receiving levels of radio waves transmitted from said mobile station;

a power level comparing section for comparing said power level with a predetermined level;

a memory for storing levels of channels determined to have a level above said predetermined level and a respective channel number; and

a level selection section for selecting a channel having a highest level of said channels determined to have a level above said predetermined level, said level selection section providing an output to said means for monitoring.

22. The system according to claim **20**, wherein said means for monitoring comprises:

a postamble monitoring/supervising section for receiving an output from said means for ranking and for monitoring postambles of a selected channel, such that a data packet is transmitted when a postamble is received by said means for monitoring; and

a collision monitoring section for determining whether a collision has occurred between said voice communication and said transmitting of data packets,

wherein, when said collision monitoring section detects a collision, said collision monitoring section provides an output to terminate said transmitting of data packets, and when no collision is detected by said collision monitoring section, said transmitting of said data packets continues to completion.

23. The system according to claim **21**, wherein said means for monitoring comprises:

a postamble monitoring/supervising section for receiving an output from said means for ranking and for monitoring postambles of a selected channel, such that a data packet is transmitted when a postamble has been received by said means for monitoring; and

a collision monitoring section for determining whether a collision has occurred between said voice communication and said transmitting of data packets,

wherein, when said collision monitoring section detects a collision, said collision monitoring section provides an output to terminate said transmitting of data packets, and when no collision is detected by said collision monitoring section, said transmitting of said data packets continues to completion.

24. A method for acquiring a channel in radio packet data communication system including a base station and a mobile station, in communication with said base station, for transmitting a voice burst, comprising:

receiving said voice burst and measuring a receiving level thereof;

ranking a receiving state based on receiving levels of radio waves transmitted from said mobile station;

selecting a channel with a predetermined receiving state based on said ranking, and monitoring the receiving state of the channel;

determining whether said mobile station is not transmitting a voice burst on the channel;

based on said determining, interrupting said channel and transmitting packet data thereover; and

when said determining determines a voice burst is transmitted on said channel during transmission of said packet data, suspending the transmission of said packet data, and identifying a second mobile station not transmitting a voice burst, thereby to obtain an idle channel for radio packet data communication.

25. The method according to claim **24**, further comprising:

when said idle channel obtained for the packet data transmission is used for voice communication before termination of packet data transmission, suspending the transmission of packet data;

monitoring for a burst which indicates termination of the voice burst; and

interrupting a channel not transmitting a voice burst.

26. The method according to claim **24**, further comprising:

when the voice burst is terminated and a silent state is determined, determining that the channel is usable, and transmitting packet data.

27. The method according to claim **26**, further comprising:

when starting of the voice burst as a talking state is obtained again, confirming a state where no burst is transmitted on the channel;

determining that said channel cannot be used; and

terminating the packet data transmission.

28. The method according to claim **24**, wherein said ranking comprises:

measuring a power level of said receiving levels of radio waves transmitted from said mobile station;

comparing said power level with a predetermined level; and

storing levels of the channels determined to have a level above said predetermined level and a respective channel number, and

wherein said selecting comprises:

selecting a channel having a highest level of said channels determined to have a level above said predetermined level.

29. The method according to claim **24**, wherein said determining comprises:

monitoring postambles of a selected channel to control transmission of packet data; and

determining whether a collision has occurred between said voice burst and said packet data,

wherein, when a collision is detected, said radio packet data transmission is terminated, and when no collision is detected, said radio packet data transmission continues to completion.

* * * * *

US005960357A

United States Patent [19]
Kim

[11] **Patent Number: 5,960,357**
[45] **Date of Patent: Sep. 28, 1999**

[54] **METHOD FOR EXECUTING AUTOMATIC CALLING FUNCTION IN A CORDLESS TELEPHONE SYSTEM**

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[73] Assignee: **SamSung Electronics Co., Ltd.**, Suwon, Rep. of Korea

[21] Appl. No.: **08/702,229**

[22] Filed: **Aug. 23, 1996**

[30] **Foreign Application Priority Data**

Aug. 23, 1995 [KR] Rep. of Korea 95-26179

[51] **Int. Cl.**^{6} **H04Q 7/30**; H04M 1/00
[52] **U.S. Cl.** **455/462**; 455/90; 379/434
[58] **Field of Search** 455/462, 550, 455/575, 90, 564, 556, 412, 413, 414, 418, 434, 435, 460, 426, 461, 458, 567, 445, 417, 416; 379/428, 433, 434, 40, 51

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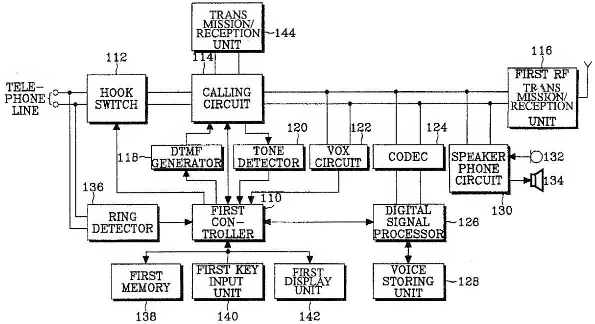
Primary Examiner—Dwayne D. Bost
Assistant Examiner—Jean A. Gelin
Attorney, Agent, or Firm—Robert E. Bushnell, Esq.

[57] **ABSTRACT**

A method for registering a telephone number for executing an automatic calling function in a cordless telephone system comprising a base station and a "flip-type" remote handset. The base station includes a hook switch for forming a calling loop, a VOX circuit for detecting a non-voice state, a tone detector for detecting a busy tone for informing the state of unavailability of the subscriber's telephone, a memory for registering a telephone number of a destination subscriber thereto and a speaker phone for establishing a telephone call through an external microphone and a speaker. The "flip-type" remote handset includes a flip switch for detecting an opening state and a closing state of a flip and a call key for a calling request. The automatic calling function is executed by: receiving a key input signal corresponding to a telephone number of destination subscriber for registration into a memory; establishing an automatic calling mode for automatically calling the destination subscriber and dialing the registered telephone number upon a calling request; and cutting off the calling when a calling end is detected and converting the automatic calling mode into a call standby state.

23 Claims, 8 Drawing Sheets



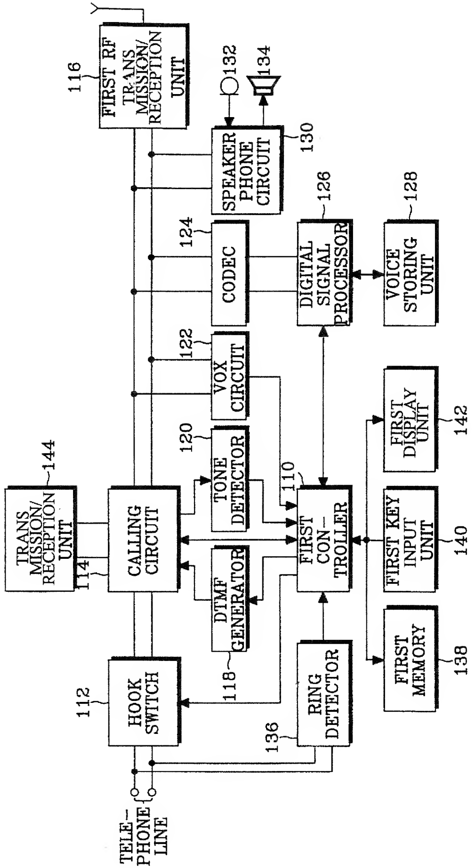

116
FIRST RF TRANS MISSION/ RECEPTION UNIT
132
134
SPEAKER PHONE CIRCUIT
130
124
CODEC
126
DIGITAL SIGNAL PROCESSOR
128
VOICE STORING UNIT
122
VOX CIRCUIT
120
TONE DETECTOR
110
FIRST CON-TROLLER
142
FIRST DISPLAY UNIT
144
TRANS MISSION/ RECEPTION UNIT
114
CALLING CIRCUIT
DTMF GENERATOR
118
140
FIRST KEY INPUT UNIT
112
HOOK SWITCH
136
RING DETECTOR
138
FIRST MEMORY
TELE-PHONE LINE

Fig. 1

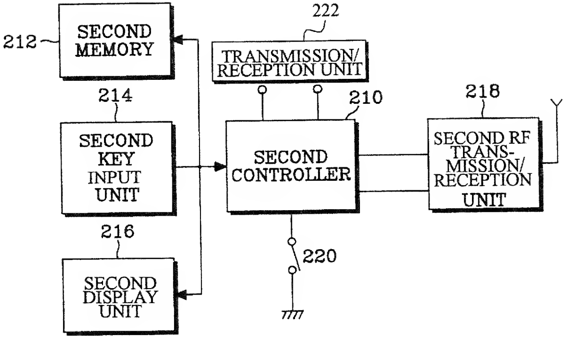

222
TRANSMISSION/
RECEPTION UNIT
212
SECOND
MEMORY
214
SECOND
KEY
INPUT
UNIT
210
SECOND
CONTROLLER
218
SECOND RF
TRANS-
MISSION/
RECEPTION
UNIT
216
SECOND
DISPLAY
UNIT
220

Fig. 2

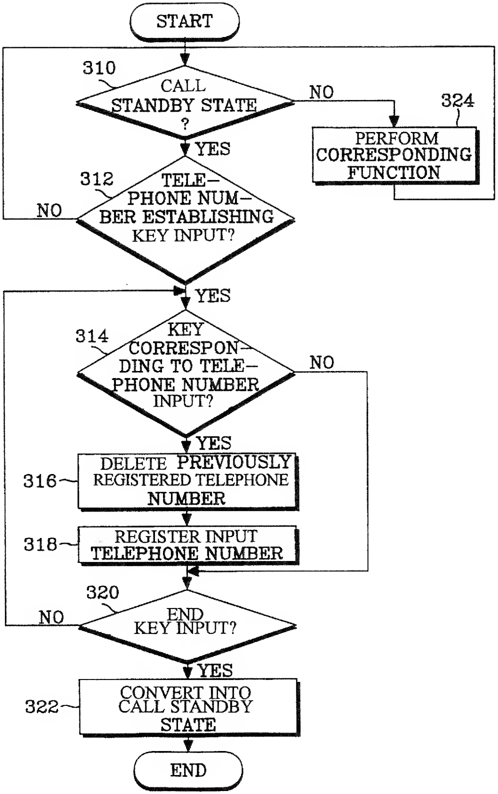

START
310
CALL STANDBY STATE ?
NO
324
PERFORM CORRESPONDING FUNCTION
YES
312
TELEPHONE NUMBER ESTABLISHING KEY INPUT?
NO
YES
314
KEY CORRESPONDING TO TELEPHONE NUMBER INPUT?
NO
YES
316
DELETE PREVIOUSLY REGISTERED TELEPHONE NUMBER
318
REGISTER INPUT TELEPHONE NUMBER
320
END KEY INPUT?
NO
YES
322
CONVERT INTO CALL STANDBY STATE
END

Fig. 3

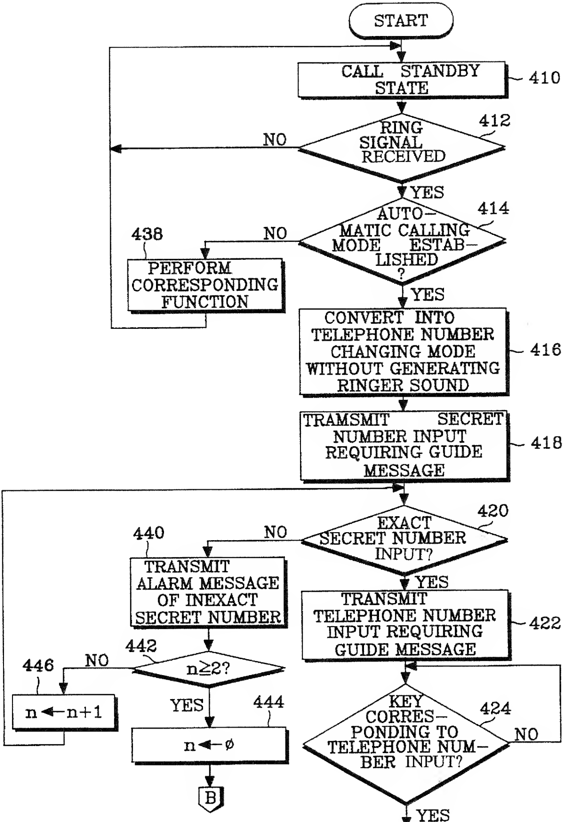

START
CALL STANDBY STATE
410
RING SIGNAL RECEIVED
412
NO
YES
AUTO-MATIC CALLING MODE ESTAB-LISHED ?
414
438
PERFORM CORRESPONDING FUNCTION
CONVERT INTO TELEPHONE NUMBER CHANGING MODE WITHOUT GENERATING RINGER SOUND
416
TRAMSMIT SECRET NUMBER INPUT REQUIRING GUIDE MESSAGE
418
EXACT SECRET NUMBER INPUT?
420
440
TRANSMIT ALARM MESSAGE OF INEXACT SECRET NUMBER
TRANSMIT TELEPHONE NUMBER INPUT REQUIRING GUIDE MESSAGE
422
442
n≥2?
446
n←n+1
444
n←ø
KEY CORRES-PONDING TO TELEPHONE NUM-BER INPUT?
424
B
A

Fig. 4A

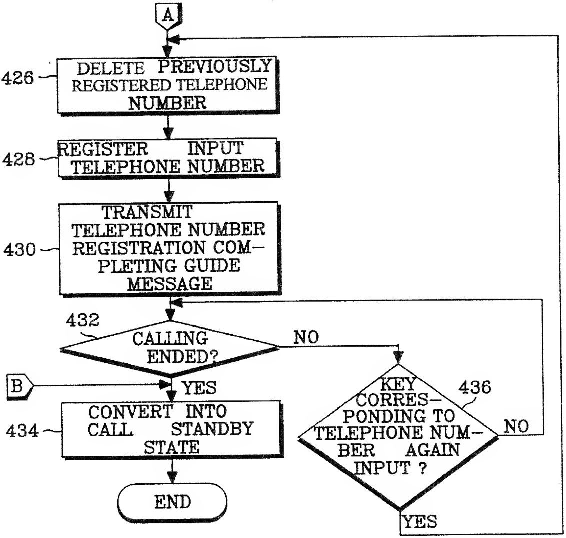

A
426
DELETE PREVIOUSLY REGISTERED TELEPHONE NUMBER
428
REGISTER INPUT TELEPHONE NUMBER
430
TRANSMIT TELEPHONE NUMBER REGISTRATION COM-PLETING GUIDE MESSAGE
432
CALLING ENDED?
NO
YES
B
434
CONVERT INTO CALL STANDBY STATE
END
436
KEY CORRES-PONDING TO TELEPHONE NUM-BER AGAIN INPUT ?
NO
YES

Fig. 4B

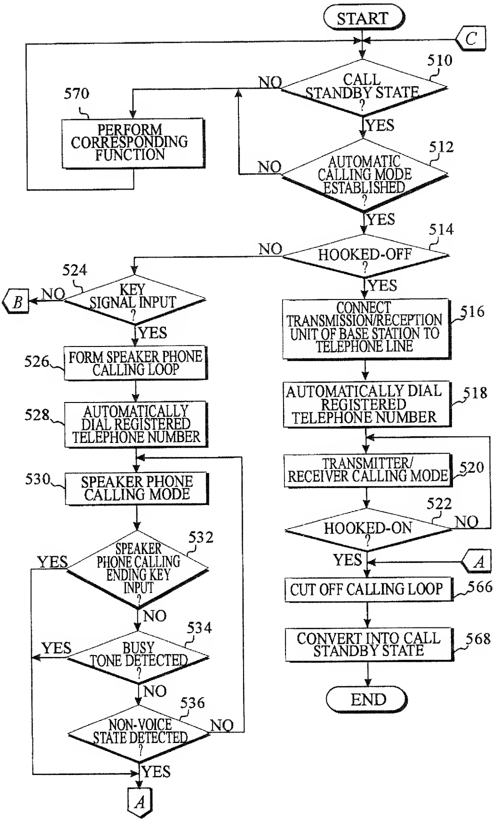

START
C
510
CALL STANDBY STATE ?
NO
YES
570
PERFORM CORRESPONDING FUNCTION
512
AUTOMATIC CALLING MODE ESTABLISHED ?
NO
YES
514
HOOKED-OFF ?
NO
YES
524
KEY SIGNAL INPUT ?
B
NO
YES
526
FORM SPEAKER PHONE CALLING LOOP
528
AUTOMATICALLY DIAL REGISTERED TELEPHONE NUMBER
530
SPEAKER PHONE CALLING MODE
532
SPEAKER PHONE CALLING ENDING KEY INPUT ?
YES
NO
534
BUSY TONE DETECTED ?
YES
NO
536
NON-VOICE STATE DETECTED ?
NO
YES
A
516
CONNECT TRANSMISSION/RECEPTION UNIT OF BASE STATION TO TELEPHONE LINE
518
AUTOMATICALLY DIAL REGISTERED TELEPHONE NUMBER
520
TRANSMITTER/ RECEIVER CALLING MODE
522
HOOKED-ON ?
NO
YES
A
566
CUT OFF CALLING LOOP
568
CONVERT INTO CALL STANDBY STATE
END

Fig. 5A

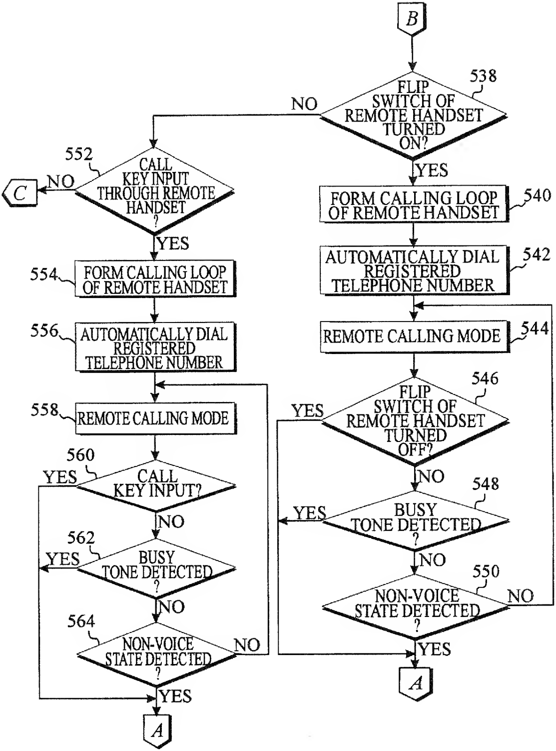

B
538
FLIP SWITCH OF REMOTE HANDSET TURNED ON?
NO
YES
540
FORM CALLING LOOP OF REMOTE HANDSET
542
AUTOMATICALLY DIAL REGISTERED TELEPHONE NUMBER
544
REMOTE CALLING MODE
546
FLIP SWITCH OF REMOTE HANDSET TURNED OFF?
YES
NO
548
BUSY TONE DETECTED ?
YES
NO
550
NON-VOICE STATE DETECTED ?
NO
YES
A
552
CALL KEY INPUT THROUGH REMOTE HANDSET ?
NO
C
YES
554
FORM CALLING LOOP OF REMOTE HANDSET
556
AUTOMATICALLY DIAL REGISTERED TELEPHONE NUMBER
558
REMOTE CALLING MODE
560
CALL KEY INPUT?
YES
NO
562
BUSY TONE DETECTED ?
YES
NO
564
NON-VOICE STATE DETECTED ?
NO
YES
A

Fig. 5B

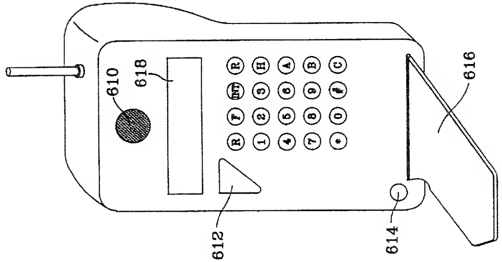


Fig. 6B

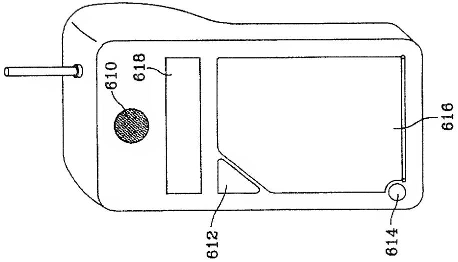


Fig. 6A

METHOD FOR EXECUTING AUTOMATIC CALLING FUNCTION IN A CORDLESS TELEPHONE SYSTEM

CROSS-REFERENCE TO RELATED APPLICATION

This application makes reference to, incorporates the same herein, and claims all benefits accruing under 35 U.S.C. §119 from an application for Method For Executing Automatic Calling Function In A Cordless Telephone System earlier filed in the Korean Industrial Property Office on Aug. 23, 1995, and there duly assigned Ser. No. 26179/1995.

BACKGROUND OF THE INVENTION

1. Technical Field

The present invention relates to a method for executing an automatic calling function in a cordless telephone system comprising a fixed base station and a "flip-type" remote handset having a flip cover, and more particularly, to a method for automatically forming a calling loop of an established telephone number when a hook switch of the base station is off-hooked, or when a dial key corresponding to a telephone number is input by a user, or when a flip cover of a remote handset is opened thereby activating a flip switch, or when a call key is input by a user.

2. Background Art

In general, conventional cordless telephone systems include a fixed base station connected to the telephone line and a remote portable handset connected to the base station by wireless. In such a telephone system, telephone calls can be made with the remote portable handset at a place away from the base station by connecting the remote portable handset to the telephone line through the base station.

One conventional housing for such a remote portable handset is a "flip-type" housing such as disclosed in U.S. Pat. No. 5,384,844 for Pivotable Housing For Hand-Held Transceiver issued to Rydbeck and U.S. Pat. No. 5,327,584 for Portable Radio Having Cover Releasing mechanism And Receive Switch Which Are Operable Together issued to Adachi et al., in which a flip cover, or lid, colloquially known as a "flip" is used to provide protection to a keypad from unintentional activation or exposure to the elements while concomitantly providing a convenient extension to the phone. When the flip is closed, a flip switch electrically connected thereto is closed and the cordless telephone system is usually in a standby mode corresponding to an "on-hook" condition. A call key remains, however, accessible to the user so that, when there is an incoming phone call and the flip is closed, the user can answer the phone call by pressing the call key without having to first open the flip. For the "flip-type" handset to perform other operating functions such as dial or redial function and intercom function, the flip has to be opened to establish an "off-hook" condition and the call key has to be activated before a dial key and function key can be activated to perform a corresponding function.

In such conventional cordless telephone system, necessity often arises for individual users who are very young such as an infant or who are very old such as an illiterate and old person and who have difficulties in memorizing telephone numbers or dialing such telephone numbers to call the same subscriber frequently or to make an emergency call to a necessary destination, and repeated dialing operations are troublesome for everyone in the daily use of the cordless telephone system. Consequently, an effective automatic calling scheme for such a cordless telephone system is necessarily required.

Generally, there are a modest number of automatic calling schemes available in the market today. For example, in U.S. Pat. No. 5,493,604 for Portable Telephone Set With Automatic Dialing Feature, Hirayama '604 provides a memory for storing dial number information of a frequently called subscriber so that, when an automatic dial function is selected, the dial number information of the frequently called subscriber is automatically sent out to the telephone network. While this automatic calling scheme has its own merit, it has been my observation that further improvements can be contemplated.

SUMMARY OF THE INVENTION

Accordingly, it is therefore an object of the present invention to provide an improved cordless telephone system comprising a base station and a "flip-type" remote handset having an automatic calling function which is capable of automatically dialing a telephone number.

It is another object to provide a method for registering a telephone number of a predetermined subscriber and changing a registered telephone number of the predetermined subscriber in a cordless telephone system for an automatic dialing function.

It is also an object to provide a method for automatically dialing a telephone number of a predetermined subscriber when a hook switch of a base station is turned off or when a key is input through a "flip-type" remote handset.

It is further an object to provide a method for automatically dialing a telephone number of a predetermined subscriber when a call key is input through a "flip-type" remote handset.

It is another object to provide a method for automatically dialing a telephone number of a predetermined subscriber upon activation of a flip switch of a "flip-type" remote handset when a flip of the "flip-type" remote handset is opened.

It is yet another object to provide a method for cutting off a calling path of a cordless telephone system when a busy tone is detected during a busy line of either a base station or a "flip-type" remote handset, or when a non-voice state is maintained during a predetermined time period.

It is also another object to provide a method for cutting off a calling path of a cordless telephone system when a hook switch of a base station is turned on during its busy line, or when a flip switch of a "flip-type" remote handset is turned off during its busy line.

These and other objects can be achieved by a method for executing an automatic calling function in a cordless telephone comprising a base station including a hook switch for forming a calling loop, a VOX circuit for detecting a non-voice state, a tone detector for detecting a busy tone, a memory for registering a telephone number of a predetermined subscriber and a speaker phone for establishing a telephone call through an external microphone and a speaker, and a "flip-type" remote handset including a flip switch for detecting an opening state and a closing state of a flip and a call key for a calling request. An automatic calling function is performed by receiving a key input signal corresponding to a telephone number of a predetermined subscriber for registration in a memory; establishing an automatic calling mode for automatically calling the predetermined subscriber and dialing the registered telephone number from the memory upon reception of a calling request to thereby establish a calling path; and cutting off the calling path when a calling end signal is detected and converting the automatic calling mode into a call standby state.

The present invention is more specifically described in the following paragraphs by reference to the drawings attached only by way of example.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete appreciation of the present invention, and many of the attendant advantages thereof, will become readily apparent as the same becomes better understood by reference to the following detailed description when considered in conjunction with the accompanying drawings in which like reference symbols indicate the same or similar components, wherein:

FIG. **1** illustrates a base station of a cordless telephone system constructed according to the principles of the present invention;

FIG. **2** illustrates a "flip-type" remote handset of a cordless telephone system constructed according to the principles of the present invention;

FIG. **3** is a flow chart illustrating a process of registering a telephone number for an automatic dialing through a base station according to the present invention;

FIGS. **4**A and **4**B are flow charts illustrating a process of changing a telephone number in the outside for executing an automatic calling according to the present invention;

FIGS. **5**A and **5**B are flow charts illustrating a process of forming a calling loop of an established telephone number according to the present invention; and

FIGS. 6A and 6B are perspective views of a "flip-type" remote handset of a cordless telephone system according to the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring now to the drawings and particularly to FIGS. **1** and **2**, which respectively illustrate a base station and a "flip-type" remote handset of a cordless telephone system constructed according to the principles of the present invention. As shown in FIG. **1**, the base station includes a first controller **110** for controlling operation of the cordless telephone system, a hook switch **112** connected to a telephone line, a calling circuit **114**, a first RF transmission/reception unit **116**, a dual-tone multi-frequency (DTMF) generator **118**, a tone detector **120**, a voice exchange (VOX) circuit **122**, a coder-decoder circuit (CODEC) **124**, a digital signal processor **126**, a voice storing unit **128**, a speaker phone circuit **130**, a microphone **132**, a speaker **134**, a ring detector **136**, a first memory **138**, a first key input unit **140**, a first display unit **142** and a transmissions/reception unit **144**. The "flip-type" remote handset as shown in FIG. **2** includes a second controller **210** for controlling operation of the cordless telephone system, a second memory **212**, a second key input unit **214**, a second display unit **216**, a transmission/reception unit **222** a second RF transmission/reception unit **218**, and a flip switch **220**. The operation of the cordless telephone system will now be described with reference to FIGS. **1** and **2** as follows.

First controller **110** generally controls the base station of the cordless telephone system. When a call key signal generated in response to depression of a call key located on the "flip-type" remote handset is received or when a flip switch **220** of the remote handset is turned on, the first controller **110** forms the calling loop of the remote handset. In addition, when the hook switch **112** is turned off, the first controller **110** forms the calling loop of a transmission/reception unit **144** of the base station. And, when dial keys

are input by the user from a first key input unit **140**, the first controller **110** forms the calling loop of a speaker phone circuit **130** of the base station.

Hook switch **112** is connected to a telephone line connected to a telephone network and is used to connect and cut-off the telephone line from a calling circuit **114** under control of the first controller **110** in order to form or terminate the calling loop. The calling circuit **114** is controlled by the first controller **110** to distribute and transmit a voice signal and all types of tone signals received from the telephone line to a corresponding DTMF generator **118** and tone detector **120**. A first RF transmission/reception unit **116** is connected to the calling circuit **114** to transmit the voice signal and the tone signals to the "flip-type" remote handset of the cordless telephone system. A DTMF s generator **118** is connected between the calling circuit **114** and the first controller **110** to convert digital data output from the first controller **110** into a DTMF signal and input the converted DTMF signal to the calling circuit **114**. A tone detector **120** is connected between the calling circuit **114** and the first controller **110** to detect the tone signals output from the calling circuit **114**, convert the detected tone signal into digital data and input the converted digital data to the first controller **110**.

VOX circuit **122** is connected between the calling circuit **114** and the first controller **110** to detect whether there is a voice signal in the busy line and provide a detected result to the first controller **110**. The CODEC **124** is connected in parallel with the VOX circuit **122** between the calling circuit **114** and the first RF transmission/reception unit **116** to convert an analog signal into digital data and input converted digital data to a digital signal processor **126**, and alternatively, to convert digital data output from the digital signal processor **126** into the analog signal and input the converted analog signal to the calling circuit **114**. The digital signal processor **126** is controlled by the first controller **110** to synthesize or analyze digital data input thereto and output from the CODEC **124** into an encoded signal. A voice storing unit **128** is connected to the digital signal processor **126** to store therein or read therefrom the encoded signal output from the digital signal processor **126**.

Speaker phone circuit **130** comprising a microphone **132** and a speaker **134** is connected to the calling circuit **114** to form a speaker phone calling loop. The microphone **132** converts the voice signal of the user into an electrical signal in a speaker phone mode and transmits the converted electrical signal to the speaker phone circuit **130**. The speaker **134** converts the electrical signal received from the speaker phone circuit **130** into the voice signal for audible reproduction. A ring detector **136** is connected between the telephone line and the first controller **110** to detect a ring signal received from the telephone line and input the detected ring signal to the first controller **110**. A first memory **138** includes a function table for containing therein all kinds of functions of the cordless telephone system. An electrically erasable programmable read-only-memory (EEPROM) may be utilized as the first memory **138** so that the first memory **138** can retain stored data even when a power supply is cut-off. A first key input unit **140** typically includes a keypad having a plurality of dial keys (e.g., 0 through 9 inclusive, a "#" key and a "*" key) and function keys for allowing the user to operate the base station of the cordless telephone system. The first display unit **142** is connected to the first controller **110** to provide a visual display of data provided from the first controller **110**. The transmission/reception unit **144** is connected to the calling circuit **114** to reproduce or transmit the voice signal through the base station in response to a general telephone call.

Second controller **210** controls the general operation of the "flip-type" remote handset. When a call key signal generated in response to depression of a call key located on the "flip-type" remote handset is received or when a flip switch **220** of the remote handset is turned on, the second controller **210** generates corresponding data to the first controller **110** of the base station. The second memory **212** includes a function table for containing therein all kinds of functions of the remote handset. Again, the EEPROM may be utilized as the second memory **212** so that the second memory **212** can retain stored data even when a power supply is cut-off A second key input unit **214** typically includes a call key and a keypad having a plurality of numeric dial keys (e.g., 0 through 9 inclusive, a "#" key and a "*" key) and function keys for allowing the user to operate the remote handset of the cordless telephone system. A second display unit **216** is connected to the second controller **210** to provide a visual display of data output from the second controller **210**. A second RF transmission/reception unit **218** is connected to the second controller **210** to transmit thereto or receive from the base station the voice signal and the tone signals under control of the second controller **210**.

Turning now to FIG. **3** which illustrates a process of registering a telephone number of a predetermined subscriber through a base station of a cordless telephone system for executing an automatic calling function according to the present invention. Generally, the process involves the steps of determining whether a telephone number establishing key is input when a base station is in a call standby state; determining whether a numeral key corresponding to a telephone number of a predetermined subscriber is input after a call standby mode is converted into a telephone number establishing mode in response to input of the telephone number establishing key; deleting a previously registered telephone number if a telephone number of a predetermined subscriber is input, and then registering the telephone number as a new number; and converting the telephone number establishing mode back into the call standby state when a call ending key is input.

FIGS. **4**A and **4**B illustrate a process of changing a telephone number of a predetermined subscriber registered in a base station of a cordless telephone system for executing an automatic calling function from an outside and remote telephone system connected to the telephone network according to the present invention. Generally, the process of changing a telephone number registered in the base station involves the steps of determining whether an automatic calling mode is established when a ring signal is received from a telephone line while the base station is in a call standby state; converting the automatic calling mode into a telephone number changing mode without generating a ringer sound, and transmitting a secret number input requiring guide message to a remote and different telephone system; determining whether an exact secret number is input from the remote telephone system in correspondence to the secret number input requiring guide message; transmitting a telephone number input requiring guide message for executing an automatic calling function when the exact secret number is input from the remote and different telephone system; determining whether a telephone number is input from the remote and different telephone system in correspondence with the telephone number input requiring guide message; deleting the previously registered telephone number if the outside telephone number is input, and then registering the input outside telephone number; transmitting the outside telephone number registration completing guide message if the registration of the outside telephone number is ended, and determining whether the calling from the remote and different telephone system is ended to convert the telephone number changing mode back into the call standby state; re-registering a numeral key corresponding to the outside telephone number when the numeral key is again input; transmitting an alarm message of inexact secret number if the inexact secret number is input in correspondence with the secret number input requiring guide message, and then counting a counter; and setting the counter as zero (0) when the counter has a counting value greater than a predetermined value, and then converting the telephone number changing mode back into the call standby state.

FIGS. **5**A and **5**B illustrate a process of forming a calling loop of an established outside telephone number according to the present invention. Generally, the process of forming such a calling loop involves the steps of determining whether an automatic calling mode is established when a base station is in a call standby state; determining whether a hook switch is off-hooked, and when the hook switch is hooked-off forming a calling loop of the base station and automatically dialing a registered outside telephone number of a predetermined subscriber to function in a transmitter/receiver calling mode; cutting off the calling loop and converting the transmitter/receiver calling mode back into a call standby mode, when the hook switch is on-hooked; forming a speaker phone calling loop and automatically dialing the registered telephone number of a predetermined subscriber to function in a speaker phone calling mode; cutting off the calling loop and converting the speaker phone calling mode back into a call standby mode, when a speaker phone calling ending key is input or when a busy tone or a non-voice state is detected; forming the calling loop of the "flip-type" remote handset and automatically dialing the registered telephone number to perform in a remote calling mode, when the automatic calling mode is established and a flip switch of the "flip-type" remote handset is turned on; cutting off the calling loop of the "flip-type" remote handset and converting the remote calling mode into the call standby state, when the flip switch is turned off or when the busy tone or the non-voice state is detected; forming the calling loop of the "flip-type" remote handset and automatically dialing the registered telephone number to perform in a remote calling mode, when the automatic calling mode is established, and determining whether a call key signal is input through the "flip-type" remote handset; cutting off the calling loop of the "flip-type" remote handset and then converting the remote calling mode into the call standby state, when the call key signal is input through the "flip-type" remote handset or when the busy tone or the non-voice state is detected.

Turning now to FIGS. **6**A and **6**B which show perspective views of a "flip-type" remote handset of the cordless telephone system constructed according to the principles of the present invention. The "flip-type" remote handset typically includes an antenna, a speaker **610** provided on an upper front surface thereof, a call key **612**, a microphone **614** provided on a lower front surface thereof a flip **616** rotatably attached to a bottom edge thereof, and a display unit **618** which is usually in a form of a liquid crystal display for providing a visual display of an operation state of the remote handset. The flip **616** as shown in FIG. **6**A is in a closed position. When the flip **616** is in a closed position, the flip **616** covers partially the keypad leaving the call key exposed and accessible to the user and thereby enabling the user to access the call key necessary to answer an incoming call in response to an alarm, e.g., ringing sound, without opening the flip **616**. When the flip **616** is in an opened position as

shown in FIG. **6**A, however, the keypad comprising a plurality of numeric dial keys, function keys such as a redial key R, a function termination key K, intercom key INT, a flash key R, a memory key 11, and one touch dial keys A, B, C is fully accessible for operation.

Refer back to FIGS. **1** and **3** in which a process of registering a telephone number of a predetermined subscriber for automatically forming the calling path will now be described in detail as follows.

First, the first controller **110** of FIG. **1** determines whether a base station is in a call standby state at step **310**. When the base station is not in a call standby state, the first controller **110** performs corresponding function at step **324**. If, on the other hand, the base station is in a call standby state, the first controller **110** determines whether a telephone number establishing key is input through the first key input unit **140** converting a call standby mode into a telephone number establishing mode at step **312**. If there is no telephone number establishing key input through the first key input unit **140** at step **312**, the first controller **110** returns to step **310** to determine whether a base station is in a call standby state. When there is a telephone number establishing key input at step **312**, however, the first controller **110** determines next whether a numeral key corresponding to a telephone number of a predetermined subscriber is input through the first key input unit **140** at step **314**. If such a numeral key corresponding to the telephone number of a predetermined subscriber is input through the first key input unit **140**, the first controller **110** deletes a conventional telephone number registered in the first memory **138** at step **316**, and then registers the telephone number of the predetermined subscriber as a new number in the first memory **138** at step **318**. Once the telephone number of the predetermined subscriber is registered in the first memory **138** at step **318**, the first controller **110** determines whether an end key is input through the first key input unit **140** for terminating the registration of a new telephone number at step **320**. When the end key is input through the first key input unit **140** at step **320**, the first controller **110** converts the base station back to a call standby state at step **322**. When there is no numeral key corresponding to the telephone number of a predetermined subscriber input through the first key input unit **140** at step **314**, however, the first controller **110** proceeds to step **320** to determine whether the end key is input in order to convert the base station back to a call standby state.

Refer back to FIGS. **1** and **4A–4B** in which a process of changing a telephone number of a predetermined subscriber registered in a base station of a cordless telephone system for executing an automatic calling function from a remote and different telephone system connected to the telephone network will now be described in detail as follows.

First, the first controller **110** determines whether a base station is in a call standby state at step **410**. When the base station is in a call standby state, the first controller **110** determines whether a ring signal is received from the telephone line at step **412**. When the ring signal is received from the telephone line at step **412**, the first controller **110** determines whether an automatic calling mode is established at step **414**. When the automatic calling mode is not established at step **414**, the first controller **110** performs corresponding function at step **438** and returns to step **410** determining whether the base station is in the call standby state.

When the automatic calling mode is established at step **414**, the first controller **110** converts such an automatic calling mode into a telephone number changing mode without generating a ringer sound according to a program stored in the first memory **138** at step **416**, and then transmits a secret number input requiring guide message read from the voice storing unit **128** through the CODEC **124** to a remote and different telephone system connected to the telephone network requesting the change in the telephone number registered in the first memory **138** for executing an automatic calling function at some later time at step **418**. Once the secret number input requiring guide message is transmitted to the remote and different telephone system at step **418**, the first controller **138** determines whether an exact secret number is input from the remote and different telephone system at step **420**. If the exact secret number is not input from the remote and different telephone system at step **420**, the first controller **110** reads an alarm message of the incorrect secret number from the voice storing unit **128** under control of the digital signal processor **126**, converts the read alarm message into a voice signal through the CODEC **124** and transmits the converted voice signal representing an alarm message of inexact secret number to the remote and different telephone system at step **440**.

After an alarm message of inexact secret number is transmitted back to the remote and different telephone system at step **440**, the first controller **110** determines whether a counter value of an internal counter reaches a predetermined value, for example, a constant two at step **442**. If the the counter value reaches the predetermined value, the counter is set as zero (0) at step **444** and then the telephone number changing mode is converted into the call standby state at step **434**. If, on the hand, the counter value is less than the predetermined value, the counter value is added by a constant value such as one at step **446**, and the first controller **110** returns to step **420** to determine whether an exact secret number is input from a remote and different telephone system connected to the telephone network.

But, if the exact secret number is received from a remote and different telephone system connected to the telephone network at step **420**, the first controller **110** controls the digital signal processor **126** to read a telephone number input requiring guide message registered to the voice storing unit **128**, convert the guide message into the voice signal through the CODEC **124** and transmit the converted voice signal representing such a telephone number input requiring guide message to the remote and different telephone system connected to the telephone network at step **422**. The first controller **110** then determines whether a numeral key corresponding to the telephone number is input from the remote and different telephone system in correspondence with the telephone number input requiring guide message at step **424**. In the case that the numeral key is input from the remote and different telephone system, the first controller **110** deletes the telephone number previously registered to the first memory **138** at step **426**, and registers a new outside telephone number input from the remote and different telephone system in the first memory **138** at step **428**. Also, in order to inform that the outside telephone number registration is ended, the first controller **110** controls the digital signal processor **126** to read a telephone number registration completing guide message registered to the voice storing unit **128**, convert the guide message into a voice signal through the CODEC **124** and transmit the converted voice signal representing such a telephone number registration completing guide message to the remote and different telephone system indicating that a new telephone number of a new subscriber is registered for future execution of an automatic calling function using the newly registered tele-

phone number. And then, the first processor **110** determines whether such a registration calling is ended at step **432**.

If the registration calling is not ended at step **432**, the first processor **110** determines whether a numeral key corresponding to the new telephone number of a new subscriber is received from the remote and different telephone system at step **436**. When a numeral key corresponding to the new telephone number is again input at step **436**, the first controller **110** again performs the telephone number registration procedure from step **426** to step **430**. If, however, the registration calling is ended at step **432**, the first controller **110** converts the telephone number changing mode into the call standby state at step **434**.

In the following, the process of forming a calling loop of an established outside telephone number will be described in detail with reference to FIGS. **1**, **2**, **5A**, **5B**, **6A** and **6B** as follows.

The first controller **110** first determines whether a base station is in a call standby state at step **510** of FIG. **5A**, and then determines whether an automatic calling mode is established at step **512**. In the case that the automatic calling mode is not established at step **512**, the corresponding function is performed at step **570**. In the case that the automatic calling mode is established at step **512**, however, the first controller **110** determines whether a hook switch **112** is off-hooked at step **514**.

When the hook switch **112** is off-hooked at step **514**, the first controller **110** controls the calling circuit **114** to connect the telephone line to the transmitting/receiving unit **144** of the base station at step **516**. The first controller **110** then reads the telephone number registered in the first memory **138** and controls the DTMF generator **118** to generate a DTMF signal corresponding to the read telephone number. That is, the first controller **110** automatically dials the registered telephone number at step **518** and converts to a transmitter/receiver calling mode to make the phone call through the transmitting/receiving unit **144** at step **520**. After the base station is in a transmitter/receiver calling mode at step **520**, the first controller **110** determines whether the hook switch **112** is on-hooked at step **522**. In the case that the hook switch **112** is on-hooked, the first controller **110** cuts off the calling loop of the transmitting/receiving unit **144** at step **566** and converts the transmitter/receiver calling mode into a call standby state at step **568**.

In the case that the automatic calling mode is established at step **512** and the hook switch **112** is not hooked-off at step **514**, the first controller **110** determines whether any numeral key is input through the first key input unit **140** at step **524**. When a numeral key is input at step **524**, the first controller **110** controls the calling circuit **114** to connect the telephone line to the speaker phone circuit **130** and thereby forming a speaker phone calling loop at step **526**. The first controller **110** then reads the telephone number registered in the first memory **138** and controls the DTMF generator **118** to thereby generate a DTMF signal corresponding to the read telephone number. In other words, the first controller **110** automatically dials the registered telephone number at step **528** and converts to a speaker phone calling mode in order to make the phone call through the speaker phone circuit **130** at step **530**.

After the base station is in a speaker phone calling mode at step **530**, the first controller **110** determines whether a speaker phone calling ending key is input through the key input unit **140** at step **532**. When the speaker phone calling ending key is input through the key input unit **140** at step **532**, the first controller **110** cuts off the calling loop of the speaker phone circuit **130** at step **566** and converts the speaker phone calling mode into a call standby state at step **568**. When the speaker phone calling ending key is not input through the key input unit **140** at step **532**, however, the first controller **110** determines whether a busy tone is detected by the tone detector **120** at step **534**. In the case that the busy tone representing a calling ending signal is detected by the tone detector **120** at step **534**, the first processor **110** also cuts off the calling loop of the speaker phone circuit **130** at step **566** and converts the speaker phone calling mode into a call standby state at step **568**. If the busy tone is not detected by the tone detector **120** at step **534**, however, the first controller **110** continues to determine whether a non-voice state is maintained during an established time from the VOX circuit **122**, and then cut off the calling loop of the speaker phone circuit **130** at step **566** and convert the speaker phone calling mode into a call standby state at step **568** when a non-voice state is detected by the VOX circuit **122**.

When a numeral key is not input at step **524**, the first controller **110** determines, at step **538**, whether a flip **616** of the "flip-type" remote handset as shown in FIGS. **6A** and **6B** is opened to thereby turn on the flip switch **220** of FIG. **2**. In the case that the flip switch **220** of the remote is turned on, the first controller **110** receives a signal from the second controller **210** of the "flip-type" remote handset and controls the calling circuit **114** at step **540**, thereby to connect the telephone line to the "flip-type" remote handset. The first controller **110** reads the outside telephone number registered in the first memory **138** at step **542**, and controls the DTMF generator **118** to thereby generate a DTMF signal corresponding to the read telephone number. In other words, the first controller **110** automatically dials the outside telephone number and converts to a remote calling mode in order to make the phone call through the "flip-type" remote handset at step **544**.

After the cordless telephone system is in a remote calling mode at step **544**, the first controller **110** determines whether the flip **616** of the "flip-type" remote handset is closed to thereby turn off the flip switch **220** at step **546**. In the case that the flip switch **220** is turned off, the first controller **110** cuts off the calling loop of the "flip-type" remote handset at step **566** and converts the remote calling mode into a call standby state at step **568**. And, in the case that the busy tone representing a calling ending signal is detected in the tone detector **120** at step **548** or that the non-voice state is maintained during the established time from the VOX circuit **122** at step **550**, the first controller **110** cuts off the calling loop of the "flip-type" remote handset at step **566** and converts the remote calling mode into a call standby state at step **568**.

Also, in the case that the flip switch **220** of the "flip-type" remote handset automatic calling mode is not turned on at step **538**, the first controller **110** determines whether a call key **612** is input through the "flip-type" remote handset of FIGS. **6A** and **6B** at step **552**. In the case that the call key **612** of the "flip-type" remote handset is input through the "flip-type" remote handset, the first controller **110** receives the signal from the second controller **210** of the "flip-type" remote handset and controls the calling circuit **114** to connect the telephone line to the "flip-type" remote handset at step **544**. At step **556**, the first controller **110** reads the outside telephone number registered in the first memory **138** and controls the DTMF generator **118** to thereby generate a DTMF signal corresponding to the read telephone number. That is, the first controller **110** automatically dials the outside telephone number and converts to a remote calling mode in order to make the phone call through the "flip-type" remote handset at step **558**.

Also, after the cordless telephone system is in a remote calling mode at step **558**, the first controller **110** determines again whether a call key **612** of the "flip-type" remote handset is input at step **560**. In the case that the call key **612** is input through the "flip-type" remote handset, the first controller **110** cuts off the calling loop of the "flip-type" remote handset at step **566** and converts the remote calling mode into a call standby state at step **568**. And, in the case that the busy tone representing a calling ending signal is detected in the tone detector **120** at step **562** or that the non-voice state is maintained during the established time from the VOX circuit **122** at step **564**, the first controller **110** cuts off the calling loop of the "flip-type" remote handset at step **566** and converts the remote calling mode into a call standby state at step **568**.

As discussed above, the method for executing an automatic calling function in a cordless telephone comprising a base station and a "flip-type" remote handset having a flip cover operable in an opened position and a closed position for partially covering a keypad leaving a call key exposed and accessible to a user according to the present invention realizes that an infant or an illiterate old person can automatically dial an outside telephone number to make the phone call when a numeral key is input or a transmitter/receiver is hang up, or when a call key of the "flip-type" remote handset is input or a flip cover is opened at the time of an emergency. Therefore, at the time of an emergency, there is an advantage in that the emergency can be quickly coped by simply and conveniently realizing the automatic calling function. Also, there is an advantage in which when a busy tone is detected or a non-voice state is maintained, a calling path is automatically cut off, thereby to eliminate an inconvenience caused by cutting the calling path by the key operation and thereby making it easy for the infant or the illiterate old to make the phone call.

While there have been illustrated and described what are considered to be preferred embodiments of the present invention, it will be understood by those skilled in the art that various changes and modifications may be made, and equivalents may be substituted for elements thereof without departing from the true scope of the present invention. In addition, many modifications may be made to adapt a particular situation to the teaching of the present invention without departing from the central scope thereof. Therefore, it is intended that the present invention not be limited to the particular embodiment disclosed as the best mode contemplated for carrying out the present invention, but that the present invention includes all embodiments falling within the scope of the appended claims.

What is claimed is:

1. A method for executing an automatic calling function in a cordless telephone system, said method comprising the steps of:

receiving a telephone number establishing key input signal corresponding to a first telephone number of a predetermined subscriber at a stationary base station of said cordless telephone system having a hook switch connected to a telephone network for registration in a memory, said cordless telephone system comprising a "flip-type" remote handset in wireless communication with said base station and having a flip cover operable in an opened position and a closed position for covering a keypad leaving a call key exposed and accessible to a user;

establishing an automatic calling mode at said base station for execution of an automatic calling function, and automatically dialing said first telephone number registered in said memory from said memory upon a calling request from one of said base station and said "flip-type" remote handset to form a calling path with said predetermined subscriber;

changing said first telephone number registered in said memory in response to said telephone number establishing key input signal and instructions from a remote communication system via said telephone network; and

cutting off said calling path when a call ending is detected, and converting said automatic calling mode into a call standby state.

2. The method of claim **1**, further comprised of said first telephone number of said predetermined subscriber being registered by:

inputting said first telephone number of said predetermined subscriber at said base station when said base station is in a telephone number registering mode; and

registering said first telephone number of said predetermined subscriber in said memory.

3. The method of claim **1**, further comprised of said first telephone number registered in said memory being changed by:

converting said automatic calling mode into a telephone number changing mode without generating a ringer sound when a calling signal is received from said remote communication system via said telephone network during said automatic calling mode, and transmitting a secret number input requiring guide message to said remote communication system via said telephone network requesting for input of a secret number;

transmitting a new telephone number input requiring guide message to said remote communication system via said telephone network requesting for input of a new telephone number for registration when an exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message;

registering a new telephone number of a different predetermined subscriber in said memory when said new telephone number is input from said remote communication system via said telephone network in correspondence with said telephone number input requiring guide message; and

transmitting a telephone number change completing guide message to said remote communication system via said telephone network indicating completion of said registration and converting said telephone number changing mode back into said automatic calling mode.

4. The method of claim **3**, further comprising the steps of:

transmitting an alarm message to said remote communication system when an inexact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message; and

deleting said first telephone number previously registered in said memory when the exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message prior to registration of the new telephone number.

5. The method of claim **1**, further comprised of said calling path being formed by:

determining a state of said hook switch of said base station in said automatic calling mode for executing

said automatic calling function to form said calling path with said predetermined subscriber; and

dialing said first telephone number of said predetermined subscriber registered in said memory when said hook switch of said base station is off-hooked, to form said calling path.

6. The method of claim 1, further comprised of said calling path being formed by:

determining whether a numeral key is input through said base station in said automatic calling mode for executing said automatic calling function to form said calling path with said predetermined subscriber; and

dialing said first telephone number of said predetermined subscriber registered in said memory when said numeral key is input through said base station, to form said calling path.

7. The method of claim 1, further comprised of said calling path being formed by:

determining whether a flip switch of said "flip-type" remote handset in wireless communication with said base station is turned on, when the flip cover is in said opened position for executing said automatic calling function to form said calling path with said predetermined subscriber, and

dialing said first telephone number of said predetermined subscriber registered in said memory, when said flip switch of said "flip-type" remote handset is turned on, to form said calling path.

8. The method of claim 1, further comprised of said calling path being formed by:

determining whether a call key signal corresponding to depression of said call key is input from said "flip-type" remote handset in wireless communication with said base station for executing

dialing said first telephone number of said predetermined subscriber registered in said memory, when said call key signal is input from said "flip-type" remote handset, to form said calling path.

9. The method of claim 5, further comprised of said calling path being cut off when said hook switch is on-hooked and then said automatic calling mode is converted into said call standby state.

10. The method of claim 6, further comprised of said calling path being cut off when a call end key is input from said base station, or when one of a busy tone and a non-voice state is detected during a predetermined time period.

11. The method of claim 7, further comprised of said calling path being cut off when said flip switch is turned off, or when one of a busy tone and a non-voice state is detected during a predetermined time period.

12. The method of claim 8, further comprised of said calling path being cut off when said call key signal is again input from said "flip-type" remote handset, or when one of a busy tone and a non-voice state is detected during a predetermined time period.

13. A method for executing an automatic calling function in a cordless telephone system connected to a telephone network, said method comprising the steps of:

receiving a first telephone number of a predetermined subscriber for registration at a stationary base station of a cordless telephone system from a remote communication system connected to said telephone network, said cordless telephone system comprising said base station having a hook switch and a memory, and a "flip-type" remote handset in wireless communication with said base station and having a flip cover operable in an opened position and a closed position for partially covering a keypad leaving a call key exposed and accessible to a user, in said memory when said base station is in a telephone number registering mode;

automatically dialing said first telephone number registered in said memory of said base station for executing said automatic calling function to establish a calling path with said predetermined subscriber, when said hook switch of said base station is off-hooked in an automatic calling mode;

changing said first telephone number registered in said memory in response to instructions from said remote communication system via said telephone network, and

cutting off said calling path and converting said automatic calling mode into a standby mode when said hook switch of said base station is on-hooked.

14. The method of claim 13, further comprised of said first telephone number registered in said memory being changed by:

converting said automatic calling mode into a telephone number changing mode without generating a ringer sound when a calling signal is received from said remote communication system via said telephone network during said automatic calling mode, and transmitting a secret number input requiring guide message to said remote communication system requesting for input of a secret number;

transmitting a new telephone number input requiring guide message to said remote communication system via said telephone network requesting for input of a new telephone number for registration, when an exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message;

registering a new telephone number of a different predetermined subscriber in said memory, when said new telephone number is input from said remote communication system via said telephone network in correspondence with said telephone number input requiring guide message; and

transmitting a telephone number change completing guide message to said remote communication system via said telephone network indicating completion of said registration and converting said telephone number changing mode back into said automatic calling mode.

15. The method of claim 14, further comprising the steps of:

transmitting an alarm message to said remote communication system when an inexact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message; and

deleting said first telephone number previously registered in said memory, when the exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message prior to registration of the new telephone number.

16. A method for executing an automatic calling function in a cordless telephone system including a base station comprising a voice circuit for detecting a non-voice state, a tone detector for detecting a busy tone, a memory for registering a telephone number thereto, and a speaker phone for establishing a telephone call through a microphone and a speaker, said method comprising the steps of:

receiving a first telephone number of a predetermined subscriber for registration in said memory;

automatically calling said predetermined subscriber by dialing said first telephone number registered in said memory when a key signal is input through said base station, to form a calling path via said speaker phone;

changing said first telephone number registered in said memory in response to instructions from a remote communication system via a telephone network: and

cutting off said calling path via said speaker phone when a call end key is input through said base station.

17. The method of claim 16, further comprised of said first telephone number registered in said memory being changed by:

converting said automatic calling mode into a telephone number changing mode without generating a ringer sound when a calling signal is received from said remote communication system via said telephone network during said automatic calling mode, and transmitting a secret number input requiring guide message to said remote communication system requesting for input of a secret number;

transmitting a new telephone number input requiring guide message to said remote communication system via said telephone network requesting for input of a new telephone number for registration, when an exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message;

registering a new telephone number of a different predetermined subscriber in said memory when said new telephone number is input from said remote communication system via said telephone network in correspondence with said telephone number input requiring guide message; and

transmitting a telephone number change completing guide message to said remote communication system via said telephone network indicating completion of said registration and converting said telephone number changing mode back into said automatic calling mode.

18. The method of claim 17, further comprising the steps of:

transmitting an alarm message to said remote communication system, when an inexact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message; and

deleting said first telephone number previously registered in said memory when the exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message prior to registration of the new telephone number.

19. The method of claim 16, further comprised of said calling path being cut off when said call key signal is again input from said "flip-type" remote handset, or when one of a busy tone is detected by said tone detector and a non-voice state is detected by said voice circuit during a predetermined time period.

20. A cordless telephone system, comprising:

a "flip-type" remote handset comprising a call key, a keypad of numeric keys and function keys, a flip cover operable in an opened position and a closed position for covering said keypad leaving said call key exposed and a flip switch operable in response to movement of said flip cover;

a base station comprising a hook switch connected to a telephone line of a telephone network, a voice circuit for detecting a non-voice state, a tone detector for detecting a busy tone and a memory for registering a telephone number thereto, said base station including a program for allowing execution of an automatic calling function by:

receiving a first telephone number of a predetermined subscriber for registration from a remote communication system via said telephone network, in said memory;

automatically dialing said first telephone number registered in said memory for executing said automatic calling function to establish a calling path with said predetermined subscriber, when said flip switch of said "flip-type" remote handset is turned on;

changing said first telephone number registered in said memory in response to instructions from said remote communication system via said telephone network: and

cutting off said calling path with said predetermined subscriber, when said flip switch of said "flip-type" remote handset is turned off.

21. The cordless telephone system of claim 20, comprised of said calling path being cut offwhen one of a busy tone is detected by said tone detector and a non-voice state is detected by said voice circuit during a predetermined time period.

22. A cordless telephone system, comprising:

a "flip-type" remote handset comprising a call key, a keypad of numeric keys and function keys, a flip cover operable in an opened position and a closed position for covering said keypad leaving said call key exposed and a flip switch operable in response to movement of said flip cover;

a base station comprising a hook switch connected to a telephone line of a telephone network, a voice circuit for detecting a non-voice state, a tone detector for detecting a busy tone and a memory for registering a telephone number thereto, said base station including a program for allowing execution of an automatic calling function by:

receiving a first telephone number of a predetermined subscriber for registration from a remote communication system via said telephone network, in said memory;

automatically dialing said first telephone number registered in said memory for executing said automatic calling function to establish a calling path with said predetermined subscriber, when said flip switch of said "flip-type" remote handset is turned on; and

cutting off said calling path with said predetermined subscriber, when said flip switch of said "flip-type" remote handset is turned off;

wherein said program for allowing execution of said automatic calling function further permits:

converting an automatic calling mode into a telephone number changing mode without generating a ringer sound when a calling signal is received from said remote communication system via said telephone network during said automatic calling mode, and transmitting a secret number input requiring guide message to said remote communication system via said telephone network requesting for input of a secret number;

transmitting a new telephone number input requiring guide message to said remote communication system

via said telephone network requesting for input of a new telephone number for registration when an exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message;

registering a new telephone number of a different predetermined subscriber in said memory when said new telephone number is input from said remote communication system via said telephone network in correspondence with said telephone number input requiring guide message; and

transmitting a telephone number change completing guide message to said remote communication system via said telephone network indicating completion of said registration and converting said telephone number changing mode to a standby mode.

23. The cordless telephone system of claim **22**, wherein said program for allowing execution of said automatic calling function further permits:

transmitting an alarm message to said remote communication system when an inexact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message; and

deleting said first telephone number previously registered in said memory when the exact secret number is input from said remote communication system via said telephone network in correspondence with said secret number input requiring guide message prior to registration of the new telephone number.

* * * * *

US006301287B1

(12) **United States Patent**
Walley et al.

(10) **Patent No.: US 6,301,287 B1**
(45) **Date of Patent: Oct. 9, 2001**

(54) **METHOD AND APPARATUS FOR SIGNAL QUALITY ESTIMATION IN A DIRECT SEQUENCE SPREAD SPECTRUM COMMUNICATION SYSTEM**

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(73) Assignee: **Conexant Systems, Inc.**, Newport Beach, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/348,491**

(22) Filed: **Jul. 7, 1999**

Related U.S. Application Data

(63) Continuation of application No. 08/568,330, filed on Dec. 6, 1995, now abandoned.

(51) **Int. Cl.**[7] **H04B 15/00**

(52) **U.S. Cl.** **375/140**

(58) **Field of Search** 340/310.06, 825.06; 375/200, 207, 206; 455/422, 437, 502

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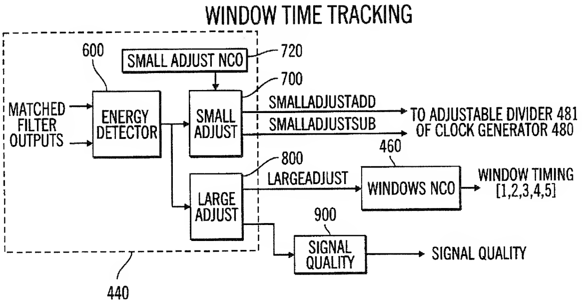
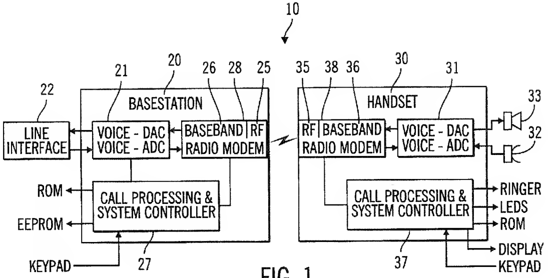
Assistant Examiner—Kevin M Burd
(74) *Attorney, Agent, or Firm*—Foley & Lardner

(57) **ABSTRACT**

A method and apparatus for estimating signal quality in a direct sequence spread spectrum communication system such as a digital cordless telephone. The method and apparatus beneficially uses data that already available from a window-tracking function used to maintain a demodulation window around the correlation data coming from the system's matched filters. The window-tracking function outputs an in-window peak value and out-window peak value for each bit interval. The signal quality is rapidly and effectively estimated by accumulating the difference between the in-window peak correlation value and the out-window peak correlation value over a period time, preferably for one frame.

15 Claims, 11 Drawing Sheets

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20
BASESTATION
21
22
LINE INTERFACE
VOICE - DAC
VOICE - ADC
26 28 25
BASEBAND RF
RADIO MODEM
ROM
EEPROM
KEYPAD
CALL PROCESSING &
SYSTEM CONTROLLER
27
30
HANDSET
35 38 36
RF BASEBAND
RADIO MODEM
31
VOICE - DAC
VOICE - ADC
33
32
CALL PROCESSING &
SYSTEM CONTROLLER
RINGER
LEDS
ROM
DISPLAY
KEYPAD
37

FIG. 1

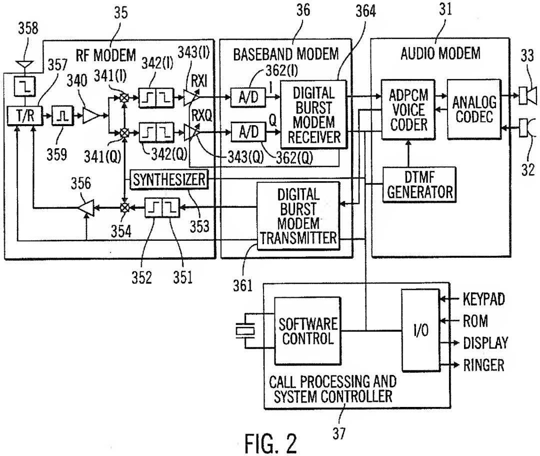

358
35
RF MODEM
357
340
341(I)
342(I)
343(I)
RXI
RXQ
T/R
359
341(Q)
342(Q)
343(Q)
SYNTHESIZER
356
354
353
352
351
36
BASEBAND MODEM
364
362(I)
I
Q
A/D
A/D
DIGITAL
BURST
MODEM
RECEIVER
362(Q)
DIGITAL
BURST
MODEM
TRANSMITTER
361
31
AUDIO MODEM
ADPCM
VOICE
CODER
ANALOG
CODEC
DTMF
GENERATOR
33
32
SOFTWARE
CONTROL
I/O
KEYPAD
ROM
DISPLAY
RINGER
CALL PROCESSING AND
SYSTEM CONTROLLER
37

FIG. 2

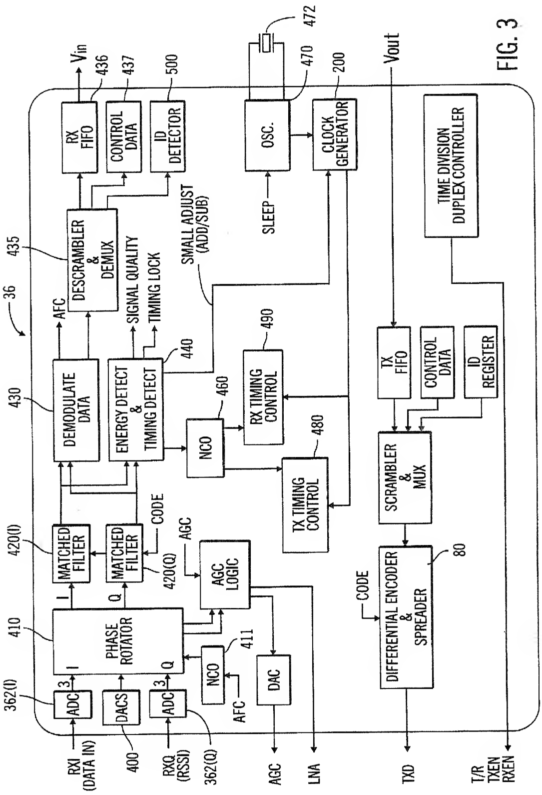

36
RX FIFO
CONTROL DATA
ID DETECTOR
OSC.
CLOCK GENERATOR
TIME DIVISION DUPLEX CONTROLLER
DESCRAMBLER & DEMUX
SMALL ADJUST (ADD/SUB)
SLEEP
DEMODULATE DATA
ENERGY DETECT & TIMING DETECT
AFC
SIGNAL QUALITY
TIMING LOCK
RX TIMING CONTROL
TX TIMING CONTROL
NCO
TX FIFO
CONTROL DATA
ID REGISTER
SCRAMBLER & MUX
MATCHED FILTER
MATCHED FILTER
CODE
AGC
AGC LOGIC
PHASE ROTATOR
DIFFERENTIAL ENCODER & SPREADER
CODE
NCO
DAC
ADC
DACS
ADC
AFC
I
Q
3
RXI (DATA IN)
RXQ (RSSI)
AGC
LNA
TXD
T/R TXEN RXEN
Vin
Vout
436
437
500
472
470
200
435
430
440
490
460
480
420(I)
420(Q)
410
411
80
362(I)
362(Q)
400

FIG. 3

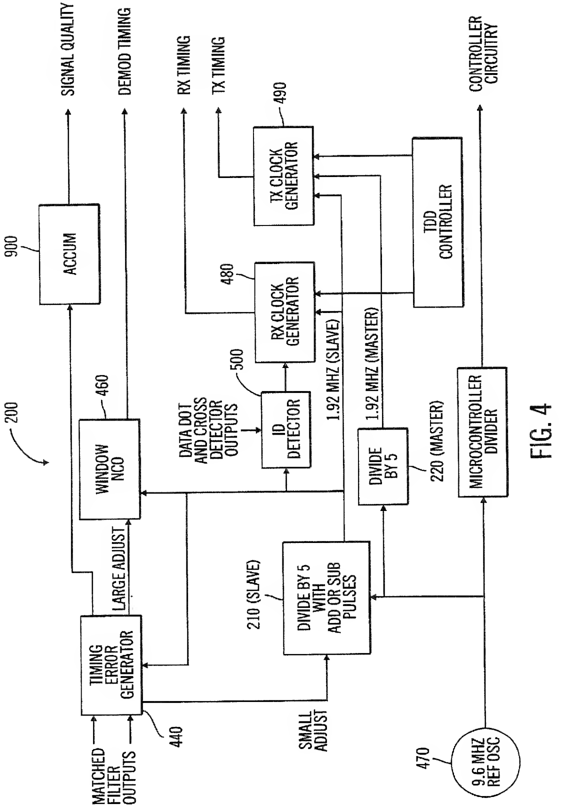
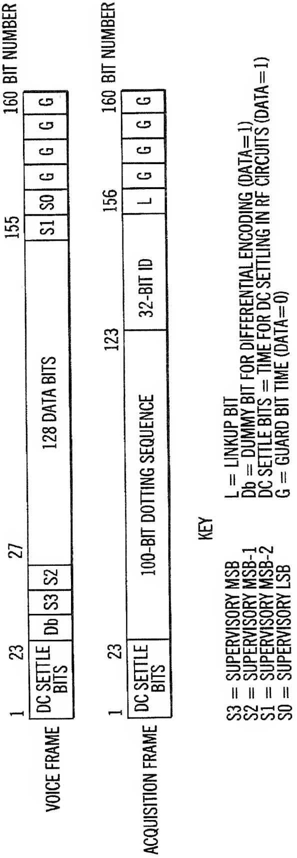

200
SIGNAL QUALITY
DEMOD TIMING
RX TIMING
TX TIMING
CONTROLLER CIRCUITRY
900
ACCUM
490
TX CLOCK GENERATOR
TDD CONTROLLER
480
RX CLOCK GENERATOR
460
WINDOW NCO
500
DATA DOT AND CROSS DETECTOR OUTPUTS
ID DETECTOR
1.92 MHZ (SLAVE)
1.92 MHZ (MASTER)
DIVIDE BY 5
220 (MASTER)
MICROCONTROLLER DIVIDER
LARGE ADJUST
210 (SLAVE)
DIVIDE BY 5 WITH ADD OR SUB PULSES
TIMING ERROR GENERATOR
MATCHED FILTER OUTPUTS
440
SMALL ADJUST
470
9.6 MHZ REF OSC

FIG. 4



VOICE FRAME
1
23
27
155
160 BIT NUMBER
DC SETTLE BITS
Db
S3
S2
128 DATA BITS
S1
S0
G
G
G
G
ACQUISITION FRAME
1
23
123
156
160 BIT NUMBER
DC SETTLE BITS
100-BIT DOTTING SEQUENCE
32-BIT ID
L
G
G
G
KEY
S3 = SUPERVISORY MSB
S2 = SUPERVISORY MSB-1
S1 = SUPERVISORY MSB-2
S0 = SUPERVISORY LSB
L = LINKUP BIT
Db = DUMMY BIT FOR DIFFERENTIAL ENCODING (DATA=1)
DC SETTLE BITS = TIME FOR DC SETTLING IN RF CIRCUITS (DATA=1)
G = GUARD BIT TIME (DATA=0)

FIG. 5

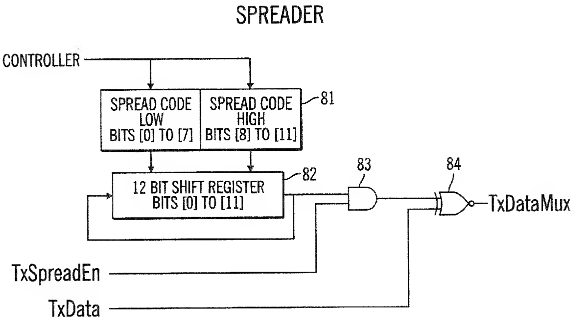

SPREADER
CONTROLLER
SPREAD CODE LOW BITS [0] TO [7]
SPREAD CODE HIGH BITS [8] TO [11]
81
12 BIT SHIFT REGISTER BITS [0] TO [11]
82
83
84
TxDataMux
TxSpreadEn
TxData

FIG. 6

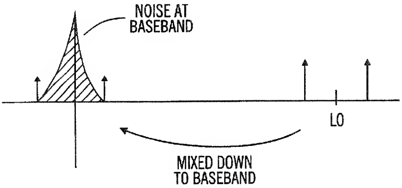

NOISE AT BASEBAND
LO
MIXED DOWN TO BASEBAND

FIG. 7

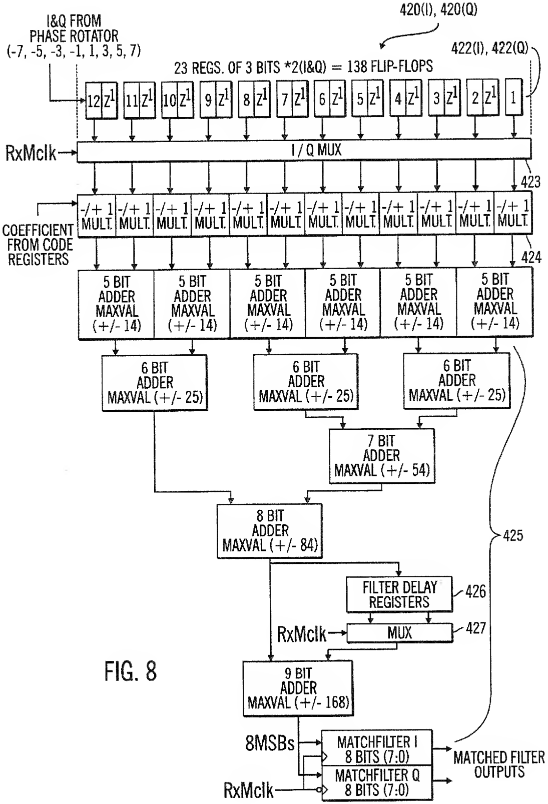

420(I), 420(Q)
I&Q FROM
PHASE ROTATOR
(-7, -5, -3, -1, 1, 3, 5, 7)
422(I), 422(Q)
23 REGS. OF 3 BITS *2(I&Q) = 138 FLIP-FLOPS
12 Z1
11 Z1
10 Z1
9 Z1
8 Z1
7 Z1
6 Z1
5 Z1
4 Z1
3 Z1
2 Z1
1
RxMclk
I / Q MUX
423
COEFFICIENT
FROM CODE
REGISTERS
-/+ 1 MULT.
424
5 BIT
ADDER
MAXVAL
(+/- 14)
6 BIT
ADDER
MAXVAL (+/- 25)
7 BIT
ADDER
MAXVAL (+/- 54)
8 BIT
ADDER
MAXVAL (+/- 84)
425
FILTER DELAY
REGISTERS
426
RxMclk
MUX
427
9 BIT
ADDER
MAXVAL (+/- 168)
8MSBs
MATCHFILTER I
8 BITS (7:0)
MATCHFILTER Q
8 BITS (7:0)
RxMclk
MATCHED FILTER
OUTPUTS

FIG. 8

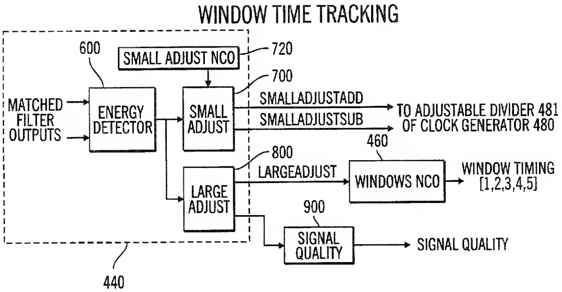

WINDOW TIME TRACKING
600
SMALL ADJUST NCO
720
700
MATCHED FILTER OUTPUTS
ENERGY DETECTOR
SMALL ADJUST
SMALLADJUSTADD
SMALLADJUSTSUB
TO ADJUSTABLE DIVIDER 481 OF CLOCK GENERATOR 480
460
800
LARGEADJUST
WINDOWS NCO
WINDOW TIMING [1,2,3,4,5]
LARGE ADJUST
900
SIGNAL QUALITY
SIGNAL QUALITY
440

FIG. 9

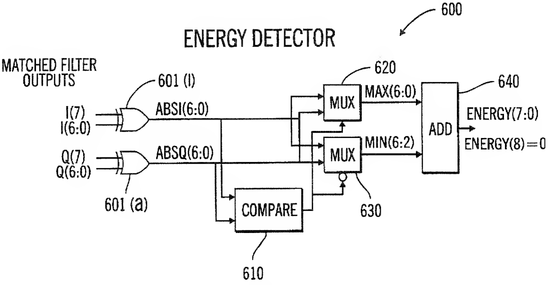

600
ENERGY DETECTOR
MATCHED FILTER OUTPUTS
601 (I)
620
640
MUX
MAX(6:0)
ABSI(6:0)
I(7)
I(6:0)
ADD
ENERGY(7:0)
MUX
MIN(6:2)
ENERGY(8)=0
Q(7)
Q(6:0)
ABSQ(6:0)
601 (a)
COMPARE
630
610

FIG. 10

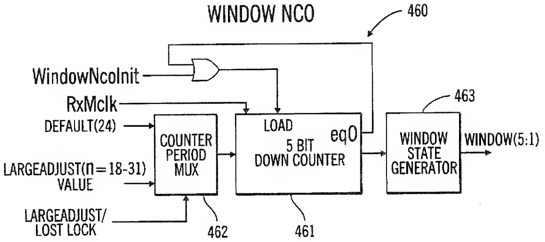
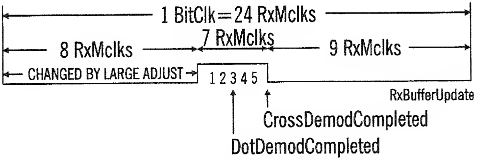


FIG. 11



INITIAL WINDOW LOCATION RELATIVE TO BIT CLOCK

FIG. 13

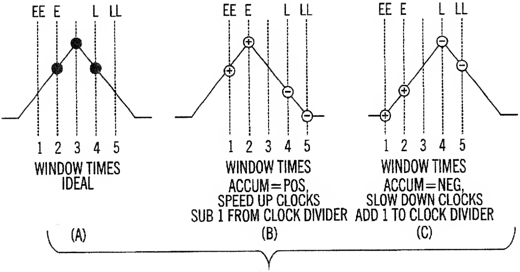


FIG. 14

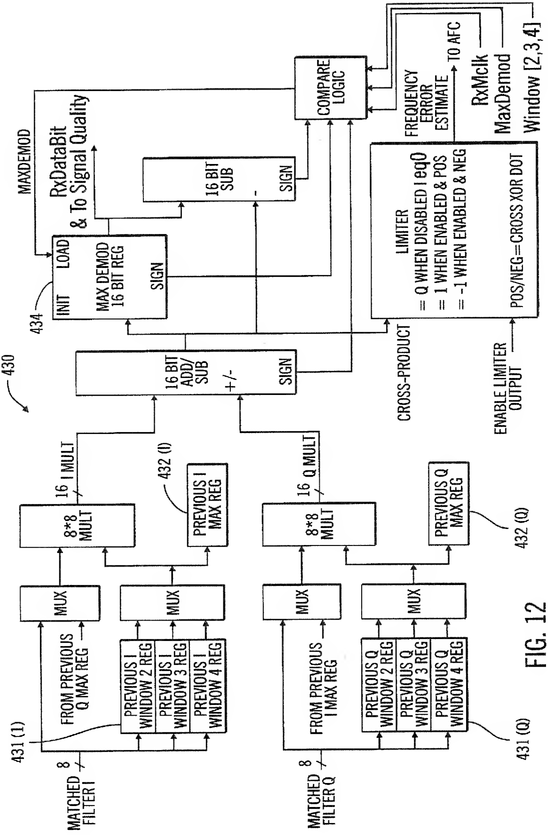

430
MATCHED FILTER I
8
431 (I)
FROM PREVIOUS Q MAX REG
PREVIOUS I WINDOW 2 REG
PREVIOUS I WINDOW 3 REG
PREVIOUS I WINDOW 4 REG
MUX
MUX
8*8 MULT
16
I MULT
432 (I)
PREVIOUS I MAX REG
MATCHED FILTER Q
8
431 (Q)
FROM PREVIOUS I MAX REG
PREVIOUS Q WINDOW 2 REG
PREVIOUS Q WINDOW 3 REG
PREVIOUS Q WINDOW 4 REG
MUX
MUX
8*8 MULT
16
Q MULT
432 (Q)
PREVIOUS Q MAX REG
16 BIT ADD/ SUB
+/-
SIGN
434
INIT
LOAD
MAX DEMOD 16 BIT REG
SIGN
MAXDEMOD
RxDataBit & To Signal Quality
16 BIT SUB
-
SIGN
COMPARE LOGIC
CROSS-PRODUCT
LIMITER
= Q WHEN DISABLED I eqQ
= 1 WHEN ENABLED & POS
= -1 WHEN ENABLED & NEG
POS/NEG=CROSS XOR DOT
ENABLE LIMITER OUTPUT
FREQUENCY ERROR ESTIMATE
TO AFC
RxMclk
MaxDemod
Window [2,3,4]

FIG. 12

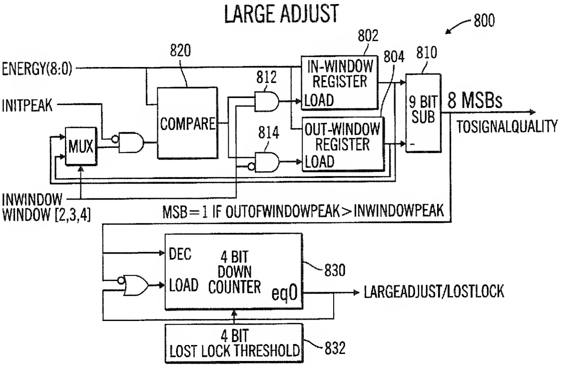

LARGE ADJUST
800
802
820
804
810
ENERGY(8:0)
IN-WINDOW REGISTER
LOAD
812
INITPEAK
COMPARE
9 BIT SUB
8 MSBs
TOSIGNALQUALITY
MUX
814
OUT-WINDOW REGISTER
LOAD
INWINDOW
WINDOW [2,3,4]
MSB=1 IF OUTOFWINDOWPEAK>INWINDOWPEAK
DEC
4 BIT DOWN COUNTER
LOAD
eq0
830
LARGEADJUST/LOSTLOCK
4 BIT LOST LOCK THRESHOLD
832

FIG. 15

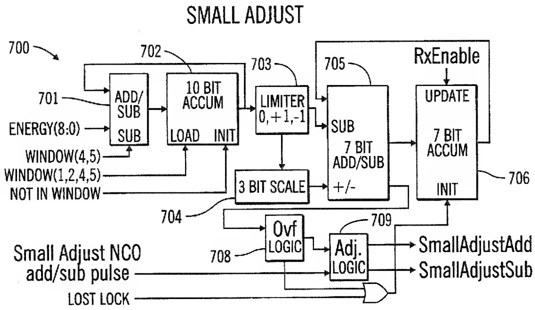

SMALL ADJUST
700
702
703
705
RxEnable
701
ADD/ SUB
SUB
10 BIT ACCUM
LOAD INIT
LIMITER 0,+1,-1
SUB
7 BIT ADD/SUB
+/-
UPDATE
7 BIT ACCUM
INIT
ENERGY(8:0)
WINDOW(4,5)
WINDOW(1,2,4,5)
NOT IN WINDOW
3 BIT SCALE
706
704
709
Ovf LOGIC
Adj. LOGIC
708
Small Adjust NCO add/sub pulse
SmallAdjustAdd
SmallAdjustSub
LOST LOCK

FIG. 16

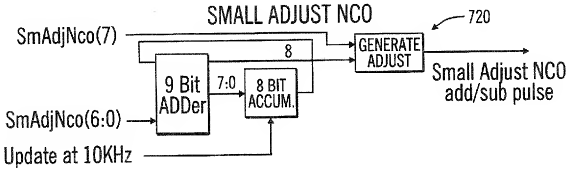

SMALL ADJUST NCO
720
SmAdjNco(7)
8
GENERATE ADJUST
9 Bit ADDer
7:0
8 BIT ACCUM.
Small Adjust NCO add/sub pulse
SmAdjNco(6:0)
Update at 10KHz

FIG. 17

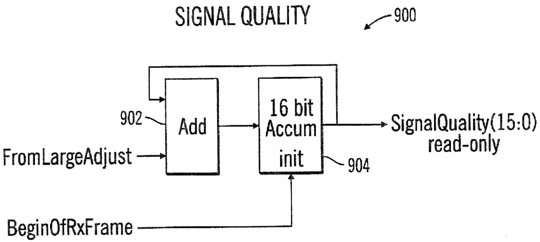

SIGNAL QUALITY
900
902
Add
16 bit Accum init
SignalQuality(15:0) read-only
FromLargeAdjust
904
BeginOfRxFrame

FIG. 18

METHOD AND APPARATUS FOR SIGNAL QUALITY ESTIMATION IN A DIRECT SEQUENCE SPREAD SPECTRUM COMMUNICATION SYSTEM

This application is a continuation of Ser. No. 08/568,330 filed Dec. 6, 1995, now abandoned.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates generally to digital cordless telephones and, more particularly, to a new and improved method and apparatus for rapidly estimating signal quality by simply accumulating the difference between the peak matched filter energy inside of a demodulation window and the peak matched filter energy outside of the window.

2. Description of Related Art

There are numerous cordless phones that use analog frequency modulation in the 46/49 MHz band. Such analog cordless phones have become very popular. Nonetheless, they suffer from range limitations, few channels, and minimal security. As a result, there is a need to provide a cordless phone product that provides a longer range without dropouts, more channels, and greater security.

Use of digital modulation and digital coding techniques offers more robust voice communication over a radio channel, although requiring greater channel bandwidth. Digital modulation also has a capture effect that greatly surpasses co-channel and adjacent channel interference, thereby providing a more noise-free conversation. Use of digital modulation encoding also allows for the addition of effective scrambling codes to greatly improve telephone security. In addition, by using the industrial, medical, and scientific (ISM) band for radio transmission and reception (902–928 MHz), increased power levels above those in the 46/49 MHz band are permitted, thus increasing the operating range. The primary FCC requirement for operating in the ISM band at the highest transmit power levels is using direct sequence spread spectrum (DSSS) or frequency hopping spread spectrum (FHSS) modulation.

DSSS modulation provides bandwidth spreading that is large compared to the bandwidth required by the information signal. A DSSS system uses a series of "chips" from a very fast code sequence for spreading an RF carrier, often by modulating the carrier using binary phase shift keying (BPSK). The receiver, of course, must duplicate the code sequence to "despread" the received signal. In order to remove the code sequence, a DSSS system generally samples the received analog signal with an A/D converter and then passes the digital signals through a matched filter at the sampling rate. The conventional DSSS system uses ordinary, 3-sample wide |E|–|L| time tracking and standard demodulation wherein the correlation "on time" peak is used for demodulation and the sample before and after the peak is used to perform a time discrimination function |E|–|L| to allow a timing NCO lock and to remove small "on time" peak timing errors. Such systems are usually called E, OT, L systems.

A hypothetical DSSS system of lowest cost and power consumption would use single sample per chip A/D conversion and matched filtering. The present inventors are unaware of any DSSS systems, however, that does this because of time tracking and demodulation problems when sampling below the Nyquist rate.

The known DSSS systems that use 3-sample wide |E|–|L| time tracking and standard demodulation typically run multiple samples per chip through the A/D conversion and matched filtering stages to effectively discriminate the correct timing. Such a brute force approach increases design complexity and fabrication cost by requiring more hardware to accommodate the multiple samples and lowers performance by requiring more compute time. The foregoing problem is particularly evident in systems that must process long code sequences for high processing gain.

The conventional DSSS modulation system is, moreover, subject to multipath fading even with multisample per chip processing because the one peak the |E|–|L| tracking system locks on may not be the best signal.

A digital communication system generally measures signal quality, in one manner or other, for use in performing system functions. The conventional methods of measuring signal quality, however, are expensive and overly complicated for the needs of a digital cordless phone.

SUMMARY OF THE INVENTION

A DSSS system according to the present invention offers a simple, robust method and apparatus for generating a signal quality value. The method and apparatus estimate the signal quality by simply determining an in-window peak correlation value from among the plurality of correlation values in a demodulation window, determining an out-window peak correlation value from among the plurality of correlation values that are not in the demodulation window, and then accumulating the difference between the in-window peak correlation value and the out-window peak correlation value over a period time.

The difference between the in-window and out-window peak correlation values is beneficially available from the system function that is used to maintain the position of the window relative to each bit interval. Accordingly, the proposed method and apparatus may generate the signal quality value with relatively little hardware or processing cost.

BRIEF DESCRIPTION OF THE DRAWINGS

The features, objects, and advantages of the present invention will become more apparent from the detailed description set forth below when taken in conjunction with the following drawings wherein:

FIG. **1** is a system-level diagram of a basestation and handset that collectively form a DCT system according to the present invention;

FIG. **2** is a functional block diagram of the RF Modem, Baseband Modem, Audio Modem, and Controller that are found in the handset, the functional blocks in the basestation being identical;

FIG. **3** is a functional block diagram of the Baseband Modem that is more detailed than that of FIG. **2**;

FIG. **4** is a functional block diagram that illustrates the generation of timing clocks in the Baseband Modem of FIG. **3**;

FIG. **5** shows the structures for two types of Tx frames that are use in the preferred digital cordless telephone embodiment, an acquisition frame (A-Frame) aud a voice frame (V-Frame);

FIG. **6** is a block diagram of the preferred spreader used in the baseband modem **36** of FIG. **3**;

FIG. **7** is a exemplary graph of power versus frequency illustrating the noise that is ordinarily introduced into baseband by direct conversion radio.

FIG. **8** is a block diagram a preferred implementation of the matched filters **420**(I), **420**(Q) shown in FIG. **3**;

FIG. 9 is a block diagram of the functions used to acquire and then maintain a Demodulation Window;

FIG. 10 is a block diagram of the Energy Detector **600** of FIG. 9;

FIG. 11 is a block diagram of the Window NCO **460** of FIG. 9;

FIG. 12 is a block diagram of the Data Demodulator of FIG. 3.

FIG. 13 is a timing diagram illustrating the initial location of the Demodulation Window relative to a bit clock;

FIGS. 14(*a*), shows the ideal relationship between the Demodulation Window and an idealized correlation output of the Matched Filters **420**(I), **420**(Q);

FIGS. 14(*b*) and 14(*c*) illustrate the relationship between the Demodulation Window and the idealized correlation output of the Matched Filters **420**(I), **420**(Q) wherein the Demodulation Window is offset in one direction or the other;

FIG. 15 is a block diagram of the Large Adjust Block **800** of FIG. 9;

FIG. 16 is a block diagram of the Small Adjust Block **700** of FIG. 9;

FIG. 17 is a block diagram of the Small Adjust NCO **720** of FIG. 9; and

FIG. 18 is a block diagram of the Signal Quality Block **900** of FIG. 9.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The following description is provided to enable any person skilled in the art to make and use the invention and sets forth the best modes contemplated by the inventors of carrying ont the invention. Various modifications, however, will remain readily apparent to those skilled in the art, since the generic principles of the present invention for providing an apparatus for estimating signal quality in a cordless direct sequence spread spectrum telephone.

Structural Overview

As shown in FIG. 1, a digital cordless telephone (DCT) **10** according to the present invention comprises a basestation **20** and at least one handset **30**. The DCT **10** is designed to provide full duplex voice communication between the handset **30** and the public switched telephone network (PSTN) (not shown) by way of an RF link through the basestation **20**.

The basestation **20** and the handset **30** both include a Radio Modem **28**, **38**, respectively, which are composed of an RF Modem **25**, **35** and a Baseband Modem **26**, **36**, respectively. The RF Modems **25**, **35** transmit digital voice and control data between the basestation **20** and the handset **30**. The basestation **20** and handset **30** also have Audio Modems **21**, **31** that respectively provide voice transport between the basestation **20** and the PSTN and between the handset **30** and a microphone **32** and a speaker **33**. Finally, the basestation **20** and the handset **30** both have a system controller **27**, **37**, respectively, for call processing and control functions. The controllers **27**, **37** function to provide the protocol for the Radio Modems **28**, **38** to allow link establishment, maintenance, and power management.

The preferred RF Modem **35** is a direct conversion transceiver. For transmit, analog wave-shaped baseband data (TXD) is modulated directly to the carrier frequency. For receive, the carrier is directly converted to analog baseband in-phase (RXI) and quadrature (RXQ) signals. The preferred DCT **10** uses frequency division multiple access (FDMA) channelization which, in the 26 MHz-wide ISM band, provides 21 frequency channels at 1.2 MHz channel spacing.

The preferred Baseband Modem **36** is a narrow-band direct sequence spread spectrum (DSSS) burst modem that supports FDMA channelization and time division duplexing (TDD). The preferred transmit and receive time slots are both 2 ms wide, providing a 4 ms Tx/Rx superframe. The preferred modulation for data is differentially encoded Binary Phase Shift Keying (BPSK), and for the spreading code is BPSK. The differential encoding of the data is done such that a change of polarity over a bit interval represents a "−1" and a continuity of polarity represents a "+1."

The DCT **10** preferably uses bipolar signalling where +1 volt signal represents a binary "**1**" and a −1 volt signal represents a binary "**0**". Each bit is preferably direct sequence spread spectrum modulated with a 12-chip spreading code. The preferred spreading code repeats on bit boundaries so that it remains "as is" when multiplied by a binary "1" (+1 volt) and is inverted when multiplied by a binary "0" (−1 volt). The preferred bit rate is 80 kHz, whereby the chip rate for the preferred 12-chip spreading code is 960 kHz (12×80 kHz).

FIG. 5 shows the preferred 160-bit frame structures for a voice frame (V-frame) and an acquisition frame (A-frame). In the V-frame, the "Db" bit (24) is always "+1" to provide an initial phase reference for decoding the differentially encoded data during reception. In the A-frame, the first 23 bits (1–23) are all DC settle bits (all "1"s), which allow for DC settling of bias circuits in the RF Modem **35**. The next 100 bits (24–123) are dotting bits (1,0,1,0, . . . repeated), used by the Baseband Modem **36** for timing acquisition and tracking during an initial reception. The next 32 bits (124–155) are an ID word, followed by an "L-bit" (156) which is used to request an RF link response. Finally, there are 4 guard bits (157–160), which account for radio propagation and circuit delays.

Referring to FIG. 2 in more detail, the preferred RF Modem **35** is shown to have a lower transmit path and an upper receive path. A common frequency synthesizer **353** provides the local oscillator frequency needed for modulation and demodulation. For transmission, the RF Modem **35** receives digital spread spectrum data TXD from the Baseband Modem **36**, and passes the data TXD through a pair of filters **351**, **352** to a modulator **354** where it directly modulates the RF carrier. The first filter **351** is preferably a 10 kHz high pass filter **351** for removing any DC contributed by bias networks and the second filter **352** is preferably a 650 kHz low-pass filter for spectral shaping. The modulated signal is then amplified using a variable output amplifier **356** and routed to a Transmit/Receive (T/R) switch **357** for transmission by an antenna **358**. For reception, the T/R switch **357** selects the receive path from the antenna **358**, filters the received signal through a band-pass filter **359** to reject out-of-band signals, and then amplifies the filtered signal with a low noise amplifier **340**. The signal then enters a pair of mixers **341**(I), **341**(Q) that down-convert it to analog in-phase (RXI) and quadrature (RXQ) spread spectrum signals. The analog spread spectrum signals RXI, RXQ are then passed through a pair of high-pass/low-pass filter blocks **342**(I), **342**(Q), respectively, to provide FDMA channel selectivity. Finally, each spread spectrum signal RXI, RXQ is amplified with programmable gain amplifiers **343** (I), **343**(Q) to bring the signal up to proper voltage levels.

With continued reference to FIG. 2, The Baseband Modem **36** is shown to have a transmitter block **361** and a pair of ADC's **362**(I), **362**(Q) that feed a receiver block **364**. During transmission, the transmitter block **361** takes analog baseband data from the Audio Modem **31**, digitizes it, differentially encodes it, combines it with a spreading code,

and then provides the resulting digital spread spectrum data TXD to the RF Modem **35** where it directly modulates the carrier for transmission. During reception, the ADCs **362(I)**, **362(Q)** convert the RF Modem's analog spread spectrum signals RXI, RXQ into digital spread spectrum data I, Q. The receiver block **364** receives the digital spread spectrum data I, Q. It then removes the spreading code to recover the digital baseband data, decodes the differentially encoded data, and provides the resulting digital baseband data to the Audio Modem **31** for acoustic reproduction on the speaker **33**.

FIG. **3** shows the major functional blocks of the Baseband Modem **36** in more detail than FIG. **2**. For transmission, the digitized voice data V-out from the Audio Modem **31** is passed through a shift register Tx FIFO, scrambled for enhanced security, and then into an Encoder/Spreader block **80** where it is differentially encoded and inversion modulated with a code sequence. FIG. **6** shows a preferred spreader wherein a preferred 12-chip code is loaded from controller registers **81** into a 12-bit shift register **82**. As the 12-chip code is shifted around the register, it is combined through AND gate **83** (for spreading enabling/disabling) to X-OR gate **84** where it modulates differentially encoded data TxData.

During reception, the in-phase and quadrature analog signals RXI, RXQ are provided to the ADC's **362(I)**. **362(Q)** for conversion to corresponding digital signals I, Q. The preferred ADC's **362(I)**, **362(Q)** sample the analog baseband signals RXI, RXQ at 1.92 MHz (2 times the chip rate) and convert such analog signals into a series of 4-bit, 2's complement, digital signals I, Q. The high and low threshold voltages of each ADC are automatically adjusted with appropriate software using DAC's **400**. The ADC's preferably implement the following quantization values to reduce the data-movement requirements (ensuring the lsb is always "1" so it may be implied) and to simplify later multiplication of the digital spread spectrum signals I, Q by ±1 (requiring only a simple inversion of the most significant three bits):

1001 (−7)
1011 (−5)
1101 (−3)
0001 (+1)
0011 (+3)
0101 (+5)
0111 (+7)

The digital signals I, Q then pass through a phase rotator **410** that is controlled by an AFC signal and NCO **411**. The phase rotator **410** adjusts the phase of the I, Q vector to improve demodulation performance by reducing frequency errors present in the signal. The details of the phase rotator **410** are not necessary for an understanding of the herein claimed invention.

Next, a pair of matched filters **420(I)**, **420(Q)** despread the digital spread spectrum signals I, Q by comparing such signals to the same spreading code that was used in the transmitter.

The Preferred Code Sequence

The chosen code is important to system operation. The code sequence is typically selected to maximize certain desired characteristics including, for example, good autocorrelation, noise immunity, Tx spectrum, and low intersymbol interference. DSSS systems often use very long code sequences (e.g. 1023 chips or more) to provide higher noise immunity by spreading the data over a wider bandwidth. In the digital cordless telephone, however, a shorter code sequence is preferred because it can be implemented at less design and fabrication cost but still offer excellent communication quality for a cordless telephone.

The FCC requires that the code sequence be ≧10 chips in length and that the system exhibit ≧10 dB in processing gain. Consequently, the present invention preferably implements a short code sequence that is close to the FCC minimum to provide a efficient, cost-effective implementation, while still providing high quality, reliable operation. Various code sequences are available as candidates. The best cope ≧10 chips long, however, is an 11-chip Barker sequence {11000100010} because it has perfect spectrum flatness and minimum off correlation values.

The cordless telephone of the present invention uses a direct conversion architecture, meaning that the RF carrier-plus-data signal is down converted directly to baseband rather than first being converted to an intermediate frequency. Since our baseband here is phone-quality voice data, the necessary bandwidth is only about 30 Hz to 3,000 Hz. As shown in FIG. **7**, however, a direct conversion architecture typically produces a great amount of low frequency noise that may deteriorate the fidelity of our voice data. The present invention counters such noise problem by ensuring that the voice data is modulated with a code sequence that has no energy at low frequencies or at DC. The invention accomplished this with an innovative modification to a standard Barker code, i.e. by adding a one more chip, a −1 or a +1, to convert the 11-chip Barker code into an even length 12-chip code that has an equal number of +1's and −1's. This 12-chip code might be called an Augmented Barker Code. The 12-chip Augmented Barker Code beneficially has no energy at low frequencies, or at DC, so that no noise is mixed down into our baseband. The noise is removed by a high pass filter, whereas the signal is left untouched in the direct conversion receiver.

An example of such a code is 111100010010. Note that there are other "good" 12-chip codes that available with varying degrees of performance in correlation values, and spectral output, such as {111000110010}. Such alternative codes have slightly different spectral shapes which allow detailed system optimization. The fundamental advantages is lower energy at low frequencies while meeting the FCC requirement of ≧10 chips/bit with minimal chips/bit.

Matched Filters

FIG. **8** shows a preferred construction for the matched filters **420(I)**, **420(Q)**. The construction shown is clocked at multiple samples per chip, namely two, given the relatively short 12-chip code sequence used in this embodiment. In operation, the digital data I, Q from the ADC's **362(I)**, **362(Q)** are simultaneously clocked into two, separate 23-long series of 3-bit registers **422(I)**, **422(Q)** at the ADC sampling rate of 1.92 MHz. As already mentioned, 1.92 MHz is 2 times the chip rate of 980 kHz. A total of 24 samples are obtained, therefore, for each bit that was modulated by our 12-chip spreading code-two samples for each chip interval.

The two matched filters **420(I)**, **420(Q)** are preferably implemented, as shown, by time-sharing a coefficient multiplier **424** and a summing network **425**. An I/Q MUX **423** is used to alternately provide the I data, and then the Q data, to the coefficient multiplier **424** and the summing network **425**. Since the Matched Filters **420(I)**, **420(Q)** are oversampled to 24 samples per bit, the filter's coefficients are also oversampled to 24 (12 chips*2 samples/chip) with zero insertion between taps.

The filter coefficients are +1 for One Code Bits, and −1 for Zero Code Bits. For example, a 12-chip spreading code of:

1 1 1 1 0 0 0 1 0 0 1 0

would result in the following coefficients used for multiplication:

+1, +1, +1, +1, −1, −1, −1, +1, −1, −1, +1, −1

The multiply operation is beneficially simplified by having already limited the digital signals I, Q to 4-bit, 2's complement values of (−7, −5, −3, −1, 1, 3, 5, 7). Multiplying by −1 simply requires an inversion of the 3 msbs and multiplication by +1 requires no change at all.

The preferred circuit of FIG. **8** further minimizes hardware by summing 24 values in 12-value increments. This 12+12 summing is effected by summing the first 12 values for I & Q, storing these values in filter delay registers **426**, and then adding such values to the next 12 values through a summing MUX **427**. Note that only 12 of the 23 values held in the shift registers **422**(I), **422**(Q), or every other one, are passed through the I/Q MUX **423** at any one time, 11 of the other 12 values being temporarily held between the first values that are passed and the 12th value coming in from the phase rotator **410**.

The maximum output of the matched filters is ±168, values that would occur only if the digital spread spectrum data I, Q from the ADC's was arriving at ±7 and all 24 samples were in perfect code correlation (24*7=168). The peak correlation values from the matched filters will typically be significantly less than the maximum, around ±120, because the ADC's will calibrate under ±7.

The outputs of the Matched Filters **420**(I), **420**(Q) are provided to a Data Demodulator Block **430** and to an Energy/Timing Detect Block **440**.

Matched Filter Outputs—Demodulation

The Data Demodulator Block **430** demodulates the differentially encoded data contained in the complex I/Q vector at times associated with so-called Window Timing Control signals that are generated by a Window NCO **460** (not shown in FIG. **3**, but see FIG. **4**). In particular, the I & Q Matched Filter Data are processed at the appropriate times using dot product and cross product demodulation on I & Q data that is exactly 1 bit clock apart, where:

Dot Product=Icurrent*Idelayed+Qcurrent*Qdelayed

Cross Product=Icurrent*Qdelayed−Qcurrent*Idelayed

The Data Demodulator Bock **430** then outputs the data through a Descrambler/Demultiplexer **435** to an Rx Buffer **436** (FIFO) for rate adaption, a Control Data Block **437** for storing supervisory data contained in V-frames, and an ID Detector **500**.

It is sufficient here to visualize the ID Detector Block **500** as monitoring the output of the Data Demodulator Block **430** in order to detect the ID word in the A-frame of FIG. **5**. The ID Detector **500** performs this detection function by simply determining if some 32-bit long sequence of bits matches the 32 bits corresponding to the predetermined ID word. In particular, the ID Detector **500** shifts in the data, repeatedly compares it with an ID Register, and generates an IDdetect signal if it matches. The IDword is used to inhibit false links from nearby transmitters or noise and also, as explained further below, to establish frame timing.

Matched Filter Outputs-Energy Detection

The matched filters **420**(I), **420**(Q) are also connected, as shown in FIG. **3**, to an Energy/Timing Detect Block **440**. As shown in FIG. **9**, the Energy/Timing Detect Block **440** comprises several suboperations including an Energy Detector **600** that drives a Small Adjust Block **700** and a Large Adjust Block **800** that are used for synchronization and tracking, and also to a Signal Quality Block **900**.

FIG. **10** show a preferred construction for the Energy Detector **600** of FIG. **9** in more detail. As shown, the Energy Detector **600** simply generates a 9-bit energy metric for use by the timing and signal quality circuits **700**, **800**, **900**. The metric is derived from the I & Q outputs from the Matched Filters **420**(I), **420**(Q) and is an estimate of the signal's envelope. The ideal energy detector would calculate the power of the I/Q vector using the formula

$$\text{Power}=(I^2+Q^2)^{1/2}$$

This formula, however, requires high compute at the Rx sample rate which, in the preferred embodiment, is 1.92 MHz. An alternative formula that requires less compute is:

$$\text{Energy Estimate}=\text{Max}(|I|,|Q|)+\tfrac{1}{4}*\text{Min}(|I|,|Q|)$$

The I and Q values will be 8-bit 2's complement numbers. Accordingly, the absolute values may be approximated by using the 7 lsbs if the number is positive and using the inverted 7 lsbs if the number is negative. The absolute value of the negative value will be off by one lsb since the addition of 1 lsb is required to complement a 2's complement number.

The preferred Energy Detector **600** of FIG. **10** implements the above Energy Estimate using a pair of X-OR gates **601**(I), **602**(Q) to obtain the absolute values of I & Q, a compare block **610** that compares the absolute values and selects the larger value with a Maximum MUX **620** and the smaller value with a Minimum MUX **630**, and an adder **640** that adds the larger value to ¼th of the smaller value.

The Energy Detector **600** outputs this Energy Estimate calculation as an 8-bit positive value which, as shown, requires a 9th bit (fixed value=0) for a 2's complement representation.

Synchronization and Tracking-Generally

Returning to FIG. **3**, one sees that the Baseband Modem **36** includes Tx Timing and Rx Timing Control Blocks **480**, **490** that are nominally driven by a system clock source consisting of a reference oscillator **470** and its associated crystal **472**. The reference oscillator **470** preferably oscillates at 9.6 MHz and drives a main clock generator circuit **200** that, in turn, provides lower frequency clocking for transmission and reception via dividers in the Tx Timing and Rx Timing Control Blocks **480**, **490**.

For transmit purposes, the Baseband Modem **36** has two timing synchronization modes: "Master" or "Slave." Either the basestation **20** or the handset **30** can be the Master, depending on which is initiating the RF link. The noninitiating unit become the Slave. If the Baseband Modem **36** is operating as the Master, then its transmit timing is derived from its own free-running clock source. If the Baseband Modem **36** is operating as a Slave, then its transmit timing is slaved to the signal received from the transmitting Master source. The Rx timing is always derived from the received signal, however, regardless of whether the Baseband Modem **36** is operating as a Master or a Slave when transmitting.

As generally suggested by FIG. **3**, the Baseband Modem **36** synchronizes Rx Timing (always Slave mode) and Tx Timing (sometimes Slave mode) by controlling the main clock generator circuit **200** with a "Small Adjusta" Add or Subtract pulse.

As shown more clearly in FIG. **4**, the clock generator **200** preferably generates the slower clocks pulses used for Tx and Rx Timing by dividing the 9.6 MHz output of the reference oscillator **470** through a pair of divide-by-5 dividers that nominally output 1.92 MHz main clocks: (1) a Slave Clock Divider **210**, and (2) a Master Clock Divider **220**. The Master Clock Divider **220** is only connected to the Tx Timing Block **490**, whereas the Slave Clock Divider **210** is

connected to both the Rx Timing and Tx Timing Blocks **480**, **490**. The Slave Clock Divider **210** is adjustable by ±1 pulse whereby it will divide by 4 or 6, instead of by 5, for one cycle. In other words, the Slave Clock Divider **210** ordinarily divides the reference oscillator's 9.6 MHz output by 5 to produce a 1.92 MHz clock. However, given a Small Adjust Add or Adjust Subtract pulse, the Slave Clock Divider **210** will temporarily require one more or one less 9.6 MHz pulse, respectively, before outputting a pulse. As a result, the output pulse train from the Slave Clock Divider **210** is either slightly advance or slightly retarded by the duration of one reference clock pulse ($1/9.6$ MHz =0.1042 uS).

Initial bit and frame timing are established in the basestation **20**, or handset **30**, by receiving the A-Frame shown in FIG. **5**. In general, the receiving device's Baseband Modem hears the A-Frame, uses the dotting sequence (1,0,1,0, . . . repeated) and a "Large Adjust" timing loop in the Large Adjust Block **800** to coarsely position the "Demodulation Window" for each Rx Frame (i.e. to establish when to periodically look at the output of the matched filters), tries to decodes the ID word that would be contained in a valid A-frame, and if it succeeds in decoding the ID word, sets its frame timing to coincide with the time the ID word occurred. Once bit and frame timing has been initialized, the receiver shifts into tracking mode for further demodulation using a "Small Adjust" timing loop in the Small Adjust Block **700**. If necessary, the receiver returns to the "Large Adjust" timing loop to maintain or reacquire synch.

The Demodulation Window (aka Window NCO)

A correlation peak from the Matched Filters **420**(I), **420**(Q) corresponds to a code alignment between the transmitter and the receiver and, as such, identifies when the I & Q signals should be sampled for code removal and demodulation. In the absence of multipath fading, the correlation peak repeats once every bit interval or, equivalently, once every 12 chip intervals, or every 24 sample intervals. The so-called "Demodulation Window," is a logical period of time that under ideal conditions would remain centered on each correlation peak.

FIG. **13** shows the initial position of the Window relative to the bit clock. If necessary, the Large Adjust Block **800** will cause the Window NCO to move the Window's position by integral ½ chip amounts, relative to the bit clock, according to a 5-bit programmable amount.

The position of the Demodulation Window relative to the bit clock is preferably maintained by a Window NCO **460** that is implemented as shown in FIG. **11**. At its heart, the Window NCO **460** is an adjustable counter **461** that is clocked at the ADC sample rate of 1.92 MHz, or two times per chip interval.

As best shown in FIG. **14**(*a*), the Window NCO's sole purpose is to generate five discrete Window Timing control signals "Window[1,2,3,4,5]" that define a Window of time that is 5-sample clock intervals wide and centered, as best as possible, on the centroid of the correlation energy. The five control signals "Window [1,2,3,4,5]" are used by the Small Adjust Block **700** (the outer four that identify times known as EE, E, L, and LL), by the Large Adjust Block **800** (the center three), and by the Data Demodulator Block **430** (the center three).

The actual correlation curve, of course, is usually quite unlike FIG. **14**(*a*) because of discrete time sampling, pulse shaping, and, moreover, because of multipath fading. The actual correlation output may have multiple peaks. The maximum data energy, therefore, is often not at the predicted center (i.e. Window[3]) from one bit interval to the next. The preferred DCT **10** overcomes this problem by demodulating each of the I/Q vectors associated with the three center values (i.e. Window[2,3,4]) and then selecting the one that contains the greatest estimated energy as best representing valid data.

As shown in FIG. **11**, the down counter **461** in the Window NCO **460** "load's" whatever value is present at the output of a Counter Period MUX **462**. It does so at initialization and every time thereafter that it reaches zero. The counter **461** is normally loaded with a default value of 24 whereby its default period is 24 sample clocks, the number of sample clocks needed to maintain the Window's nominal Large Adjust position relative to the bit clock. However, if the Window NCO **460** receives a LargeAdjust/Lost Lock signal at its Counter Period Mux **462**, then a different LargeAdjust value will be loaded the next time the counter **461** reaches zero. The Demodulation Window will be repeatedly repositioned in this fashion until the peak In-Window energy once again exceeds the peak Out-of-Window energy.

The location of the Window relative to the Bit Clock (80 kHz) is preferably initialized at the beginning of each Rx frame. After the Large Adjust Block **800** has positioned the Window, the Small Adjust Block **700** will track the Tx timing by indirectly moving the Window during the frame, in small amounts, by adjusting the main Rx clock.

Demodulation as Related to the NCO Window

Having generally explained the Window NCO **460**, we now turn FIG. **12** to present the preferred construction and operation of the Data Demodulator Block shown as block **430** in FIG. **3**. The Data Demodulator Circuit **430** of FIG. **12** implements the dot product and cross product demodulation discussed earlier. Significantly, the Data Demodulator Circuit **430** may be controlled to perform such demodulation on only the center sample or, more preferably, on the 3 center samples (using Registers **431**, **432** and Window[2,3,4,] timing signals), after which the values are compared and the one with the largest absolute value is saved in a Max Demod register **434** for output as an RxDataBit.

Initial Synchronization/Acquisition-Large Adjust Loop

As mentioned above, the initial bit timing is accomplished with the A-frame's dotting sequence (1,0,1,0, . . . repeated). As shown in FIG. **9**, the Energy Detector **600** provides the Large Adjust Block **800** with an energy value, twice per chip. On an initial acquisition, the peak energy of the signal would not be inside of the Demodulation Window and the Large Adjust Block **800**, detecting this, would attempt to bring the signal into the Window.

FIG. **15** shows the Large Adjust Block **800** in more detail. The Large Adjust Block **800** looks at the peak estimated signal energy inside the Demodulation Window and compares it with the peak estimated signal energy outside of the Demodulation Window.

The Large Adjust Block **800** receives Window[2,3,4] timing signals from the Window NCO of FIG. **11**. The energy signal received at each of the successive Window[2,3,4] times is compared to the value already stored in the In-Window Register **802** by a Comparator **820**. If the incoming energy value is larger, the Comparator **820** causes the In-Window Register **802** to "load" it in place of the existing energy value, the final energy value being the peak value in the Window. A similar process is performed to load the peak energy value found in sample intervals located outside of the Window (21 in the preferred embodiment) into the Out-Window Register **804**. The value in the Out-Window Register **804** is then subtracted from the value in the In-Window Register **802** to determine whether or not the peak energy was larger inside of or outside of the Window.

When the peak energy inside the Window is smaller than the peak energy outside the Window, an error bit is set that decrements a down counter **830**. A single timing error bit is generated by taking only the msb of the subtractor **810**, the msb being a "0" if the subtraction result was positive and a "1" if it was negative. A programmable threshold of 0 to 15 is set via a 4-bit Lost Lock Threshold register **832**. When the programmable threshold is exceeded, the filter is cleared and the Large Adjust Block **800** outputs a "LargeAdjust/LostLock" signal that, as suggested by FIG. **11**, is used to advance the Demodulation Window by a programmable LargeAdjust Value, such as u=18–31.

Initial Synchronization/Acquisition-Small Adjust Loop |EE|+|E|−|L|−|LL | Processing

As shown in FIG. **9**, the Energy Detector **600** also provides an Energy value to the Small Adjust Block **700**. The Small Adjust Block **700** advances or retards the Main Clock Divider **210** of FIG. **4** by ±1 system clock pulses as described earlier.

The Small Adjust Block **700** determines whether or not the Demodulation Window has drifted from center as shown in FIGS. **14**(*b*) or **14**(*c*) for a predetermined number of bit intervals and, if so, generates an appropriate SmallAdjustAdd or SmallAdjustSub signal that moves the Demodulation Window toward an aligned position as shown in FIG. **14**(*a*).

FIG. **16** shows the preferred implementation of the Small Adjust Block **700**. The Small Adjust Block adds the energy value for the first two clocks of the Window (Window[1,2], aka EE and E) and subtracts the energy value for the last two clocks of the Window[4,5], aka L and LL). The energy value at the center (Window[3]) is not used. The preferred Small Adjust Block **700** uses an Adder/Subtractor **701** and a 10 bit accumulator **702** to perform the just-described |EE|+|E|−|L|−|LL| process. The result available at the output of the accumulator **702** is then processed by a Limiter **703** to determine whether it is negative, positive, or zero. The output of the Limiter **703** is used to control a Small Adjust Accumulator **706** by using an Adder/Subtractor **705** to add or subtract a programmable 3-bit Small Adjust Scale **704** to the accumulator value. If the Limiter **703** output is negative (positive), the Small Adjust Scale **704** is subtracted (added).

When the Adder/Subtractor **705** overflows, or underflows, then one clock pulse is added, or subtracted, from the Main Clock Divider **210**. The Adder/Subtractor **705** drives an Overflow Logic Block **708** which, in turn, drives an Adjustment Logic Block **709** that generates an appropriate SmallAdjustAdd signal or SmallAdjustSub signal.

A Small Adjust NCO **720**, shown in FIG. **17**, also drives the Adjustment Logic Block **709** of FIG. **16**. The Small Adjust NCO **720** is used to continuously small adjustments to approximate actual time drifts in the system due to clock errors. This is needed in addition to the small adjustments made during Rx Demodulation (when the timing errors are large) because, otherwise, timing errors of large magnitude would accumulate during transmission.

Signal Quality

An important feature for a system is to quickly and accurately detect when the signal is bad for muting audio signals. The standard calculation methods use estimated Signal-to-Noise Ratio (SNR) or bit error rate (BER) estimates with CRC coding. These methods, however, are overly complicated and often require extra overhead in the form of parity bits and associated processing.

The present invention proposes a very simple, but-effective, signal quality estimator that beneficially uses data that is already available from the matched filters **420**(I), **420**(Q), and as already processed by another system function, the Large Adjust Block **800**. As shown in FIG. **9**, the Large Adjust Block **800** communicates with a Signal Quality Block **900** which, in turn, outputs a Signal Quality signal that is calculated as follows.

The SNR is beneficially estimated according to the present invention by simply taking the peak matched filter output energy outside of a timing window that surrounds the desired signal (corresponding roughly to the Noise floor) and subtracting this value from the peak energy inside the timing window that contains the signal. The difference signal is integrated during an Rx Frame and made available to the system controller **37** for muting, when appropriate. This new approach to estimating signal quality can be described mathematically, as follows:

SNR=Σ(Peak MF Energy In Window)−(Peak MF Energy Outside of Window)

This SNR estimate is acceptable for use with short code sequences. For a short code sequence, and with appropriate Window sizes, the SNR estimate is a good monatomic indicator of the Baseband Modems's Bit Error Rate (BER) for all useable range from 10% BER to 0.01% BER. As to "appropriate" Window sizes, the SNR estimate of the present invention works best with Window sizes wherein the number of in-window samples are small as compared to out-of-window, or total, samples.

FIG. **18** shows a block diagram for the preferred Signal Quality Block **900** wherein the peak in-window/out-of-window difference data for each bit interval, provided by the Large Adjust Block **600**, is accumulated over an Rx Frame using a 16 bit accumulator **904** and an adder **902**, to generate the desired SignalQuality signal.

ID Word-Frame Timing

As mentioned above, the ID word is used to establish initial frame timing. During receive, the Large Adjust and Small Adjust timing loops in the Energy/Timing Detect block **440** lock onto the energy output of the matched filters **420**(I), **420**(Q). When a sequence of demodulated data transitions matches the ID word, the system responds and adjusts the frame timing counters to a know state.

Those skilled in the art will appreciate that various adaptations and modifications of the just-described preferred embodiment can be configured without departing from the scope and spirit of the invention. Therefore, it is to be understood that, within the scope of the appended claims, the invention may be practiced other than as specifically described herein.

What is claimed is:

1. A signal quality circuit for a direct sequence spread spectrum communication system comprising:

peak value determining means for determining an in-window peak correlation value from among the plurality of correlation values in a demodulation window and for determining an out-window peak correlation value from among the plurality of correlation values that are not in the demodulation window; and

estimating means for estimating a signal-to-noise ratio and providing a signal quality value based on a difference data between the in-window peak correlation value and the out-window peak correlation value by accumulating the difference data for each of a plurality of bit intervals over a frame interval.

2. A signal quality circuit for a direct sequence spread spectrum communication system comprising:

peak value determining means for determining an in-window peak correlation value from among the plurality of correlation values in a demodulation win-

dow and for determining an out-window peak correlation value from among the plurality of correlation values that are not in the demodulation window; and

estimating means for estimating a signal-to-noise ratio and providing a signal quality value based on a difference data between the in-window peak correlation value and the out-window peak correlation value by accumulating the difference data for each of a plurality of bit intervals over a frame interval wherein the estimating means comprises a means for providing a subtraction result by subtracting the out-window peak correlation value from the in-window peak correlation value for each bit interval and the estimating means further comprises a means for accumulating the subtraction results for a plurality of bit intervals to generate the signal quality value.

3. The signal quality circuit of claim **2** wherein the accumulating means accumulates the subtraction results over a frame interval.

4. A direct sequence spread spectrum communication system for receiving an analog spectrum baseband signal that was formed by directly modulating a carrier with the chips of a code sequence at a chip rate comprising:

A/D converting means for converting the analog spread spectrum baseband signal to corresponding digital spread spectrum baseband data;

matched filter means for removing the code sequence by correlating the digital spread spectrum data with the code sequence, said matched filter means outputting correlation values;

large adjust means for coarsely positioning a demodulation window that contains a plurality of correlation values that are in the vicinity of an ideal peak correlation value, the demodulation window having a nominal period of one bit interval to generally remain centered on the peak correlation value output by the matched filter means;

peak value determining means for determining an in-window peak correlation value from among the plurality of correlation values in the demodulation window and for determining an out-window peak correlation value from among the plurality of correlation values that are not in the demodulation window; and

estimating means for estimating a signal-to-noise ratio and providing a signal quality value based on a difference data between the in-window peak correlation value and the out-window peak correlation value.

5. The direct sequence spread spectrum communication system of claim **4** wherein the estimating means provides the signal quality value by accumulating the difference data for each of a plurality of bit intervals over a frame interval.

6. The direct sequence spread spectrum communication system of claim **4** further comprising a means for subtracting the out-window peak correlation value from the in-window peak correlation value for each bit interval to produce the difference data used by the estimating means.

7. The direct sequence spread spectrum communication system of claim **6** wherein the estimating means further comprises a means for accumulating the subtraction results for a plurality of bit intervals to generate the signal quality value.

8. A method of estimating signal quality in a direct sequence spread spectrum communication system that is receiving an analog spread spectrum signal that was formed by directly modulating a carrier with a code sequence, the method comprising the steps of:

converting the analog spread spectrum signal to corresponding digital data;

removing the code sequence by correlating the digital spread spectrum data with the code sequence;

outputting correlation values;

positioning a demodulation window that contains a plurality of correlation values that are used for demodulation;

determining an in-window peak correlation value from among a plurality of correlation values in the demodulation window and for determining an out-window peak correlation value from among a plurality of correlation values that are not in the demodulation window; and

subtracting the out-window peak correlation value from the in-window peak correlation value as a signal quality value.

9. The method of claim **8** further comprising the step of accumulating the signal quality value over a plurality of bit intervals.

10. A signal quality circuit for a direct sequence spread spectrum communication system comprising:

a circuit for accepting a Barker code, said circuit having a peak value determining means for determining, using said Barker code, an in-window peak correlation value from among the plurality of correlation values in a demodulation window and for determining an out-window peak correlation value from among the plurality of correlation values that are not in the demodulation window; and

an estimating means for estimating a signal-to-noise ratio and providing a signal quality value based on a difference data between the in-window peak correlation value and the out-window peak correlation value by accumulating difference data for each of a plurality of bit intervals over a frame interval.

11. A signal quality circuit for a direct sequence spread spectrum communication system comprising:

a circuit for accepting a Barker code, said circuit having a peak value determining means for determining, using said Barker code, an in-window peak correlation value from among the plurality of correlation values in a demodulation window and for determining an out-window peak correlation value from among the plurality of correlation values that are not in the demodulation window; and

an estimating means for estimating a signal-to-noise ratio and providing a signal quality value based on a difference data between the in-window peak correlation value and the out-window peak correlation value wherein the estimating means comprises a means for providing a subtraction result by subtracting out the out-window peak correlation value from the in-window peak correlation value for each bit interval and, the estimating means further comprises a means for accumulating the subtraction results for a plurality of bit intervals to generate the signal quality value.

12. The signal quality circuit of claim **11** wherein the accumulating means accumulates the subtraction results over a frame interval.

13. A method of estimating signal quality in a direct sequence spread spectrum communication system that is receiving an analog spread spectrum signal that was formed by modulating a carrier with a Barker code sequence, the method comprising the steps of:

converting the analog spread spectrum signal to corresponding digital data;

removing the Barker code by correlating the digital spread spectrum data with the Barker code;

outputting correlation values;

positioning a demodulation window that contains a plurality of correlation values that are used for demodulation;

determining an in-window peak correlation value from among a plurality of correlation values in the demodulation window and determining an out-window peak correlation value from among a plurality of correlation values that are not in the demodulation window; and

subtracting the out-window peak correlation value from the in-window peak correlation value as a signal quality value.

14. The method of claim 13 wherein the Barker code in an Augmented Barker Code.

15. The method of claim 13 further comprising the step of accumulating the signal quality value over a plurality of bit intervals.

* * * * *

X. APPENDIX C - RELATED PROCEEDINGS

NONE